DEFENSE INFORMATION SYSTEMS AGENCY
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IN REPLY REFER TO: Joint Interoperability Test Command (JTE) **10 Jul 14**

MEMORANDUM FOR DISTRIBUTION

SUBJECT: Joint Interoperability Certification of the Cisco Unified Communications Manager (UCM) Assured Services Session Initiation Protocol (AS-SIP) to International Telecommunication Union – Telecommunication Standardization Sector (ITU-T) H.323 Gateway with Software Release 8.6

References: (a) DoD Directive 4630.05, "Interoperability and Supportability of Information Technology (IT) and National Security Systems (NSS)," 5 May 2004
(b) Department of Defense Instruction 8100.04, "DoD Unified Capabilities (UC)," 9 December 2010
(c) through (e), see Enclosure 1

1. References (a) and (b) establish Defense Information Systems Agency (DISA), Joint Interoperability Test Command (JITC), as the responsible organization for interoperability test certification.

2. The Cisco UCM AS-SIP to H.323 Gateway with Software Release 8.6 is hereinafter referred to as the System Under Test (SUT). The SUT meets all of its critical interoperability requirements and is certified for joint use within the Defense Information Systems Network (DISN) as an AS-SIP to ITU-T H.323 Gateway. The SUT meets the critical interoperability requirements set forth in Reference (c), using test procedures derived from Reference (d). Any new discrepancy noted will be evaluated for impact on the existing certification. These discrepancies will be adjudicated to the satisfaction of DISA via a vendor Plan of Action and Milestones (POA&M), which will address all new critical Test Discrepancy Reports (TDRs) within 120 days of identification. No other configurations, features, or functions, except those cited within this document, are certified by JITC. This certification expires upon changes that could affect interoperability, but no later than three years from the date of the UC Approved Products List (APL) memorandum.

3. This finding is based on interoperability testing conducted by JITC, review of the vendor's LoC, DISA adjudication of open test discrepancy reports (TDRs), and DISA Certifying Authority (CA) Recommendation. Interoperability testing was conducted by JITC, Fort Huachuca, Arizona, from 6 through 10 May 2013. Review of the vendor's LoC was completed on 10 May 2013. DISA adjudication of outstanding TDRs was completed on 13 August 2013. Information Assurance (IA) testing was conducted by DISA-led IA test teams and the results are published in a separate report, Reference (e). Enclosure 2 documents the test results and describes the tested network and system configurations.

4. Table 1 provides a UC APL product summary. Table 2 provides the SUT interface interoperability status and Table 3 provides the Capability Requirements (CR) and Functional Requirements (FR) status. The threshold CR/FRs for AS-SIP to ITU-T H.323 Gateway components are established by Section 5.3.2.7.5 of Reference (c) and were used to evaluate the interoperability of the SUT. Enclosure 3 provides a detailed list of the interface, capability, and functional requirements.

Table 1. UC APL Product Summary

SUT (AS-SIP to H.323 Gateway) (See note 1.)	Release
Unified Communications Server (B-Series 5108) with **B200M2** Blade with ESXi 5.1 (See note 2.)	8.6.1 20011-4
UCM Publisher	8.6.1 20011-4
UCM Subscriber	8.6.1 20011-4
Interworking Gateway running on an ISRG2 (**3945**, 3945E, 3925, 3925E)	15.2(4)M3
TelePresence Video Communications Server	X8.1.1
Adaptive Security Appliance **5510**, 5505, 5520, 5540, 5585-SSP10, 5585-SSP20, 5585-SSP30, 5585-SSP40	8.4(3)

NOTES:
1. Components bolded and underlined were tested by JITC. The other components in the family series were not tested; however, they utilize the same software and similar hardware and JITC analysis determined them to be functionally identical for interoperability certification purposes and they are also certified for joint use.
2. Supported hardware configurations can be found by selecting the “Cisco Unified Communications on the Cisco Unified Computing System” link at the following URL: www.cisco.com/go/swonly.

LEGEND:

APL	Approved Products List	JITC	Joint Interoperability Test Command
AS-SIP	Assured Services – Session Initiated Protocol	SUT	System under Test
H.323	Standard for multi-media communications on packet-based networks	UC	Unified Capabilities
ISRG2	Integrated Service Router Generation 2	UCM	Unified Communications Manager

Table 2. SUT Interface Interoperability Status

Interface	Applicability	UCR 2008, Change 3 Reference	Threshold CR/FR (See note 1.)	Status	Remarks
AS-SIP – H.323 Gateway-to-UC WAN Appliance Interface					
10Base-X	C (See note 2.)	5.3.2.7.5.1	1-3, 5, 6	Met	The SUT met the critical CRs and FRs with the following IEEE standard: 802.3i (10BaseT).
100Base-X	C (See note 2.)	5.3.2.7.5.1	1-3, 5, 6	Met	The SUT met the critical CRs and FRs with the following IEEE standard: 802.3u (100BaseT).
1000Base-X	C (See note 2.)	5.3.2.7.5.1	1-3, 5, 6	Met	The SUT met CR and FRs with the following IEEE standard: 802.3ab (1000BaseT).
AS-SIP – H.323 Gateway-to-NMS Interface					
10/100 Mbps	R	5.3.2.7.5.3.7	4, 6	Partially Met	See note 3.

NOTES:
1. The SUT high-level CR and FR ID numbers depicted in the Threshold CRs/FRs column can be cross-referenced in Table 3. These high-level CR/FR requirements refer to a detailed list of requirements provided in Enclosure 3.
2. The UCR 2008, Change 3, does not specify minimum interfaces for an AS-SIP –H.323 Gateway.
3. The UCR 2008 Change 3, section 5.3.2.7.5.3.7, states that the AS-SIP – H.323 Gateway must provide a 10/100-Mbps Ethernet interface to the DISA NMS, separate from the local management interface. The SUT supports only one physical interface. DISA adjudicated this as minor with the condition of fielding to be included in the vendor’s deployment guide stipulating that local NM access and remote NM access will be provided via a single interface.

Table 2. SUT Interface Interoperability Status (continued)

LEGEND:

802.3ab	1000BaseT Gbps Ethernet over twisted pair at 1 Gbps (125 Mbps)	IEEE	Institute of Electrical and Electronics Engineers
802.3i	10BaseT Mbps over twisted pair	H.323	Standard for multi-media communications on packet-based networks
802.3u	Standard for carrier sense multiple access with collision detection at 100 Mbps	Mbps	Megabits per second
AS-SIP	Assured Services Session Initiation Protocol	NM	Network Management
C	Conditional	NMS	Network Management System
CR	Capability Requirement	R	Required
DISA	Defense Information Systems Agency	SUT	System Under Test
FR	Functional Requirement	UC	Unified Capabilities
Gbps	Gigabits per second	UCR	Unified Capabilities Requirements
ID	Identification	WAN	Wide Area Network

Table 3. SUT CR and FR Status

CR/FR ID	Capability/Function	Applicability (See note 1.)	UCR Reference	Status
1	**AS-SIP – H.323 Gateway General Requirements**			
	Call request from H.323 UC signaling platform	Required	5.3.2.7.5.1.2	Met
	Initial routine AS-SIP invite from the WAN SS/MFSS	Required	5.3.2.7.5.1.3	Met
	Process initial precedence AS-SIP invites from the WAN SS/MFSS via the EBC	Required	5.3.2.7.5.1.4	Met (See note 2.)
	AS-SIP – H.323 Gateway support for VoIP and Video Signaling Interfaces	Required	5.3.2.7.5.3.1	Met
	CCA requirements.	Required	5.3.2.7.5.3.1.1	Met
	Role of IWF within the CCA	Required	5.3.2.7.5.3.1.1.1	Met
	Resource priority header	Required	5.3.2.7.5.3.1.2	Met
	Mapping of Telephony Number into SIP URI	Required	5.3.2.7.5.3.1.3	Met
	Support for audio and video codecs	Required	5.3.2.7.5.3.2	Met
	Product Quality Factors	Required	5.3.2.7.5.3.8	Met (See note 3.)
2	**SAC**			
	The AS-SIP – H.323 Gateway must implement call count thresholds for voice sessions and for video sessions in order to perform SAC.	Required	5.3.2.7.5.1.1	Met
	SAC	Required	5.3.2.7.5.3.1.4.1	Met
	Directionalization	Conditional	5.3.2.7.5.3.1.4.2	Not Supported
	Code Blocking	Required	5.3.2.7.5.3.1.4.3	Met
	Configuration of total voice call thresholds and total video call thresholds	Required	5.3.2.7.5.3.1.4.4	Met
	Configuration of outbound voice call thresholds, inbound voice call thresholds, outbound video call thresholds, and inbound video call thresholds.	Conditional	5.3.2.7.5.3.1.4.5	Not Supported
	SAC enforcement	Required	5.3.2.7.5.3.1.4.6 5.3.2.7.5.3.1.4.7 5.3.2.7.5.3.1.4.8	Met

Table 3. SUT CR and FR Status (continued)

CR/FR ID	Capability/Function	Applicability (See note 1.)	UCR Reference	Status
3	**AS-SIP – H.323 Gateway Management Function**			
	Section 5.3.2.17.1, Voice and Video Network Management Domain	Required	5.3.2.7.5.3.5	Met
	Section 5.3.2.17.2, General Management Requirements	Required	5.3.2.7.5.3.5	Partially Met (See note 4.)
	The AS-SIP – H.323 Gateway must support one pair of Ethernet management interfaces where one management interface is for communication with a local EMS and one management interface is for communication with a remote EMS. In addition, the AS-SIP – H.323 Gateway must support at least one additional Ethernet interface for carrying signaling and media streams for VVoIP traffic.	Required	5.3.2.7.5.3.5	Not Met (See note 5.)
	Section 5.3.2.17.3.1, Fault Management	Required	5.3.2.7.5.3.5	Partially Met (See note 6.)
	Section 5.3.2.17.3.2.1, Read-Write Access to CM Data by the RTS EMS	Required	5.3.2.7.5.3.5	Met
	Section 5.3.2.17.3.4.1, Near-Real-Time Network Performance Monitoring	Required	5.3.2.7.5.3.5	Partially Met (See note 7.)
	Section 5.3.2.17.3.4.2, Remote Network Management Commands (the LSC requirements apply to the AS-SIP – H.323 Gateway with the exception of Section 5.3.2.17.3.4.2.14, PEI/GEI Origination Capability Control)	Required	5.3.2.7.5.3.5	Met
	Section 5.3.2.17.3.5, Security Management	Required	5.3.2.7.5.3.5	Met
	Requirement 5.3.2.18.1, NM Requirements for CE Routers and EBCs	Required	5.3.2.7.5.3.5	Not Met (See note 8.)
	Section 5.3.2.18.2, Management Requirements for the ASAC (use these requirements for SAC only)	Required	5.3.2.7.5.3.5	Partially Met (See note 9.)
4	**Interface to the DISA NMS**			
	The AS-SIP – H.323 Gateway must provide a 10/100 Mbps interface to the DISA NMS IAW section 5.3.2.4.4	Required	5.3.2.7.5.3.7	Not Met (See note 5.)
5	**IPv6**			
	Product Requirements for NA/SS	Required	5.3.5.4	Partially Met (See notes 10, 11, 12, 13.)
6	**Information Assurance**			
	The AS-SIP – H.323 Gateway must meet IA requirements IAW UCR 2008, Change 3, Section 5.4 for a media gateway.	Required	5.3.2.7.5.3.3	Met (See note 14.)

NOTES:

1. The annotation of 'required' refers to a high-level requirement category. The applicability of each sub-requirement is provided in Enclosure 3. The system under test does not need to provide conditional requirements. However, if a capability is provided, it must function according to the specified requirements.
2. The SUT meets the requirement by diverting the INVITE to the attendant when calls above ROUTINE are placed from the AS-SIP enclave to the H.323 non Command and Control (C2) enclave.
3. The UCR states the AS-SIP to H.323 Gateway can meet either Medium Availability or High Availability. High Availability is supported for the AS-SIP interface; however, the H.323 interface does not meet the High Availability product quality factors. Therefore, the SUT is certified as Medium Availability.
4. The SUT partially complies with the general management requirements. The SUT meets the data reporting requirements; however, it does not use SNMP or eXtensible Markup Language (XML) to do so. DISA adjudicated this as minor with the condition of fielding to be included in the vendor's deployment guide stipulating an alternative procedure to manage the gateway.
5. The SUT does not provide an NM interface to the DISA NMS. The SUT supports only one physical interface. DISA adjudicated this as minor with the condition of fielding to be included in the vendor's deployment guide stipulating that local NM access and remote NM access will be provided via a single interface.
6. The SUT partially complies with the fault management requirements. Fault reporting is not performed via SNMP. DISA adjudicated this as minor with the condition of fielding to be included in the vendor's deployment guide stipulating an alternative procedure to manage the gateway.

Table 3. SUT CR and FR Status (continued)

NOTES (continued):
7. The SUT partially complies with performance monitoring capability per the reference. This capability is available using (other) existing UCM tools. Cisco is unable to verify SUT capabilities for EMS interoperability. DISA adjudicated this as minor with the condition of fielding to be included in the vendor's deployment guide stipulating an alternative procedure to manage the gateway.
8. The SUT must comply with the following NM requirements: Faults will be reported IAW RFCs 1215 and 3418. Standard CM and PM information shall be presented IAW RFCs 1213 and 3418. DISA adjudicated this as minor with the condition of fielding to be included in the vendor's deployment guide stipulating an alternative procedure to manage the gateway.
9. The UCR requirement states that the SUT must permit reading of the VoIP, video, and other session counts (as applicable) from the VVoIP EMS. Call counts can be read from SUT management application; however, the vendor does not know if they can be read by VVoIP EMS. DISA adjudicated this as minor with the condition of fielding to be included in the vendor's deployment guide stipulating an alternative procedure to manage the gateway.
10. The SUT meets the requirements for IPv4/IPv6 dual stack with the following minor exception. The SUT does not support dual stack for H.323. DISA adjudicated this as minor. There is no mature standard for dual stack with H.323 protocol. The SUT is certified for IPv4 only on the H.323 side of the gateway. The SUT will be fielded with IPv4 on the H.323 side of the gateway.
11. The SUT VCS does not comply with Multicast Listener Discovery (MLD) as described in RFC 2710. DISA adjudicated this as minor. The SUT will be fielded with IPv4 on the H.323 side of the gateway.
12. The SUT partially complies with assigning the DSCP tagging requirements. The SUT VCS can be assigned any DSCP value 0-63, however this value is assigned to all types of packets to include OA&M, Signaling and Media. DISA accepted the vendor's PoA&M and adjudicated this as minor.
13. The SUT supports IPv6 Neighbor Discovery and SLAAC in accordance with RFCs 2461 and 2462 and not RFCs 4861 and 4862 as required by the UCR. During testing, there were no interoperability discrepancies related to IPv6 Neighbor Discovery or SLAAC. DISA accepted the vendor's PoA&M and adjudicated this as minor.
14. Security is tested by DISA-led Information Assurance test teams and the results published in a separate report, Reference (e).

LEGEND:

ASAC	Assured Services Admission Control
AS-SIP	Assured Services Session Initiation Protocol
CCA	Call Connection Agent
CE	Customer Edge
CM	Configuration Management
CR	Capability Requirement
DISA	Defense Information Systems Agency
DSCP	Differentiated Services Code Point
EBC	Edge Boundary Controller
EMS	Element Management System
FR	Functional Requirement
GEI	Generic End Instrument
IA	Information Assurance
IAW	in accordance with
ID	Identification
IPv4	Internet Protocol version 4
IPv6	Internet Protocol version 6
IWF	Inter-working Function
H.323	Standard for multi-media communications on packet-based networks
Mbps	Megabits per second
MFSS	Multi-function Softswitch
NA/SS	Network Appliance/Simple Server
NM	Network Management
NMS	Network Management System
OA&M	Operations, Administration, and Maintenance
PEI	Proprietary End Instrument
PM	Performance Management
PoA&M	Plan of Action and Milestones
RFC	Request for Comments
RTS	Real Time Services
SAC	Session Admission Control
SIP	Session Initiation Protocol
SLAAC	Stateless Auto Address Configuration
SNMP	Simple Network Management Protocol
SS	Softswitch
SUT	System Under Test
UC	Unified Capabilities
UCM	Unified Communications Manager
UCR	Unified Capabilities Requirements
URI	Uniform Resource Indicator
VCS	Video Communication Server
VoIP	Voice over Internet Protocol
VVoIP	Voice and Video over Internet Protocol
WAN	Wide Area Network

5. JITC distributes interoperability information via the JITC Electronic Report Distribution (ERD) system, which uses Sensitive but Unclassified IP Data (formerly known as NIPRNet) e-mail. Interoperability status information is available via the JITC System Tracking Program (STP). STP is accessible by .mil/.gov users at https://stp.fhu.disa.mil/. Test reports, lessons learned, and related testing documents and references are on the JITC Joint Interoperability Tool (JIT) at https://jit.fhu.disa.mil/. Due to the sensitivity of the information, the Information Assurance Accreditation Package (IAAP) that contains the approved configuration and

deployment guide must be requested directly from the Unified Capabilities Certification Office (UCCO), e-mail: disa.meade.ns.list.unified-capabilities-certification-office@mail.mil. All associated information is available on the DISA UCCO website located at http://www.disa.mil/Services/Network-Services/UCCO.

6. The JITC point of contact is Mr. Edward Mellon, commercial telephone (520) 538-5159, DSN telephone 879-5159, FAX DSN 879-4347; e-mail address edward.a.mellon.civ@mail.mil; mailing address Joint Interoperability Test Command, ATTN: JTE (Mr. Edward Mellon) P.O. Box 12798, Fort Huachuca, AZ 85670-2798. The UCCO tracking number for the SUT is 1305001.

FOR THE COMMANDER:

3 Enclosures a/s

for RIC HARRISON
Chief
Networks/Communications and UC Portfolio

Distribution (electronic mail):
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Joint Staff J-6, JCS
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ISG Secretariat, DISA, JTA
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NSG Interoperability Assessment Team
DOT&E, Netcentric Systems and Naval Warfare
Medical Health Systems, JMIS IV&V
HQUSAISEC, AMSEL-IE-IS
UCCO

ADDITIONAL REFERENCES

(c) Office of the Assistant Secretary of Defense, "Department of Defense Unified Capabilities Requirements 2008, Change 3," September 2011
(d) Joint Interoperability Test Command, "Defense Switched Network Generic Switch Test Plan (GSTP), Change 2," 2 October 2006
(e) Joint Interoperability Test Command, "Information Assurance (IA) Information Assurance (IA) Assessment of Cisco Unified Communications Manager (CUCM) 8.6 (Tracking Number 1305001)," Draft

CERTIFICATION TESTING SUMMARY

1. SYSTEM TITLE. Cisco Unified Communications Manager (UCM) Assured Services Session Initiation Protocol (AS-SIP) to International Telecommunication Union – Telecommunication Standardization Sector (ITU-T) H.323 Gateway with Software Release 8.6; hereinafter referred to as the System Under Test (SUT).

2. SPONSOR. Headquarters United States Army Information Systems Engineering Command (HQUSAISEC), Mr. Steve Pursell, USAISEC ELIE-ISE-ES, Building 53301, Fort Huachuca, Arizona 85613, e-mail: steven.d.pursell.civ@us.army.mil.

3. SYSTEM POC. Cisco Systems Global Certification Team (GCT), 7025-2 Kit Creek Road, Research Triangle Park, North Carolina 27709, e-mail: certteam@cisco.com.

4. TESTER. Joint Interoperability Test Command (JITC), Fort Huachuca, Arizona.

5. SYSTEM DESCRIPTION. The AS-SIP - H.323 Gateway interfaces to the enclave Edge Boundary Controller (EBC) in both the signaling plane and the bearer plane and is responsible for interworking AS-SIP voice and video signaling with the voice and video signaling of the H.323 Unified Communications (UC) signaling platform. The AS-SIP – H.323 Gateway is responsible for interworking Unified Capabilities Requirements (UCR)-compliant voice and video media packets with the voice and video media packets supported by the H.323 UC signaling platform's Internet Protocol (IP) End Instruments (EIs). Interoperability of UC features and services other than non-assured voice and video services is outside the scope of the required functionality for the AS-SIP – H.323 Gateway and is not a part of AS-SIP - H.323 Gateway SUT interoperability testing. The SUT consists of the components listed in the subparagraphs below.

UCM 8.6 (Cisco Unified Computing System [UCS] B200 M2 Server). The B200 M2 server is a full blade width server that provides computing, networking, storage access, and virtualization. Supported hardware configurations can be found by selecting the "Cisco Unified Communications on the Cisco Unified Computing System" link at the following URL: www.cisco.com/go/swonly. The following components were configured on the server as virtual machines:

- **Video Communication Server (VCS).** The VCS provides advanced TelePresence applications and session management to all standards-compliant infrastructure and management solutions as well as gatekeeper services, enabling interoperability between H.323 end devices. VCS is a critical part of Cisco's unified call-control solution.

- **UCM Publisher (Pub).** The UCM provides Voice over Internet Protocol (VoIP) call processing, call admission control, messaging, identification, authentication, and security services to users. It provides high availability via server redundancy and database replication and supports failover to subordinate servers in the event of hardware failure.

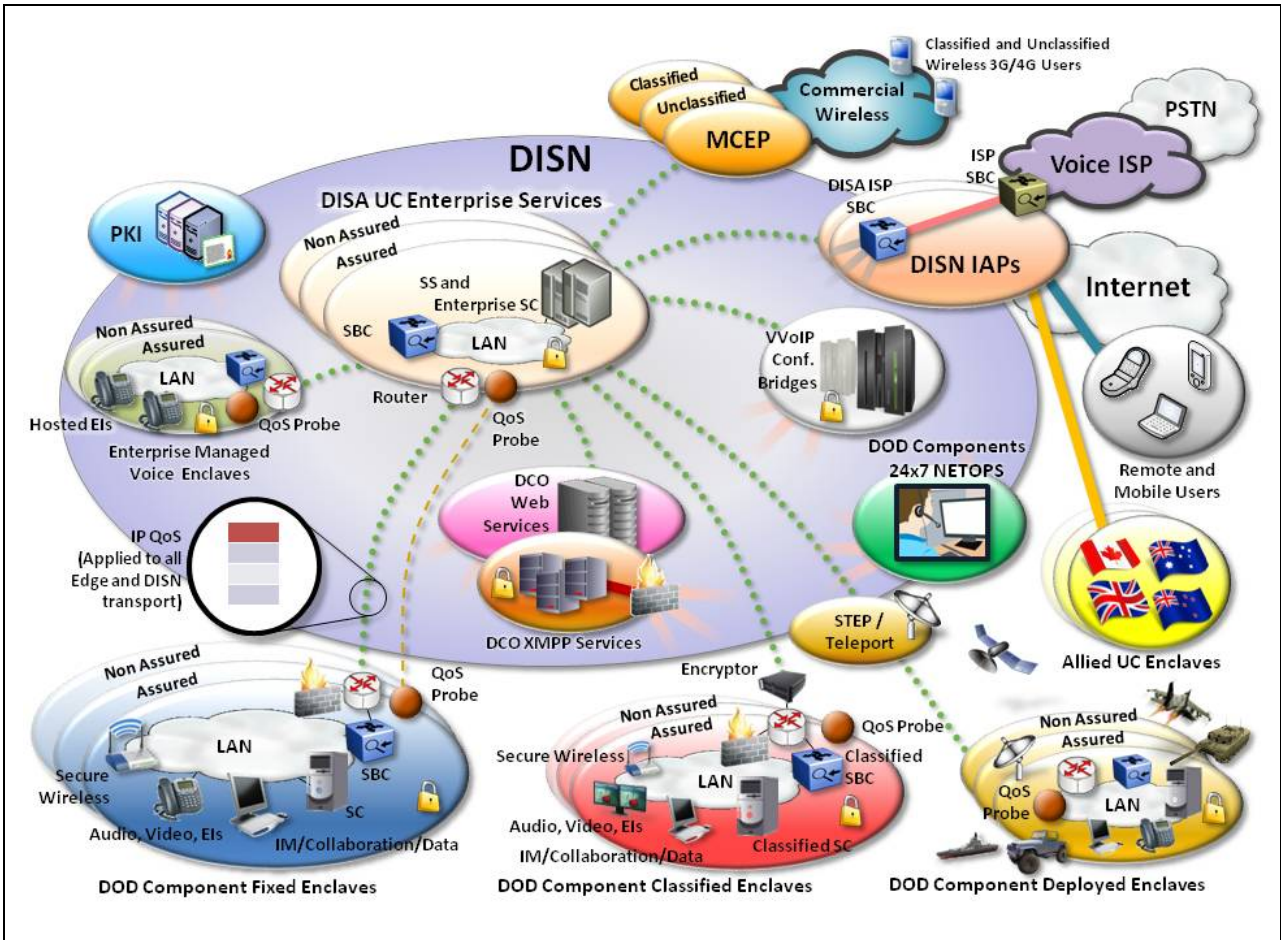
- **UCM Subscriber (Sub).** The UCM provides VoIP call processing, call admission control, messaging, identification, authentication, and security services to users. It provides high availability via server redundancy and database replication and supports failover to subordinate servers in the event of hardware failure.

3945 Interworking Gateway (IWG) (Primary and Secondary). The IWG is an intelligent UC network border element that provides Session Border Controller (SBC) functions that help enable End-to-End (E2E) IP-based transport of voice, video, and data between independent unified communications networks. It is deployed as part of the AS-SIP - H.323 gateway and provides critical Session Initiation Protocol (SIP) message normalization functionality that allows for interoperability between components within the UC architecture.

Adaptive Security Appliance (ASA) 5510. The ASA component is included in the solution to provide Common Access Card (CAC) support via Virtual Private Network (VPN) access.

Management Workstation (site-provided). A Security Technical Implementation Guide (STIG)-compliant, CAC-enabled workstation used to access the SUT components.

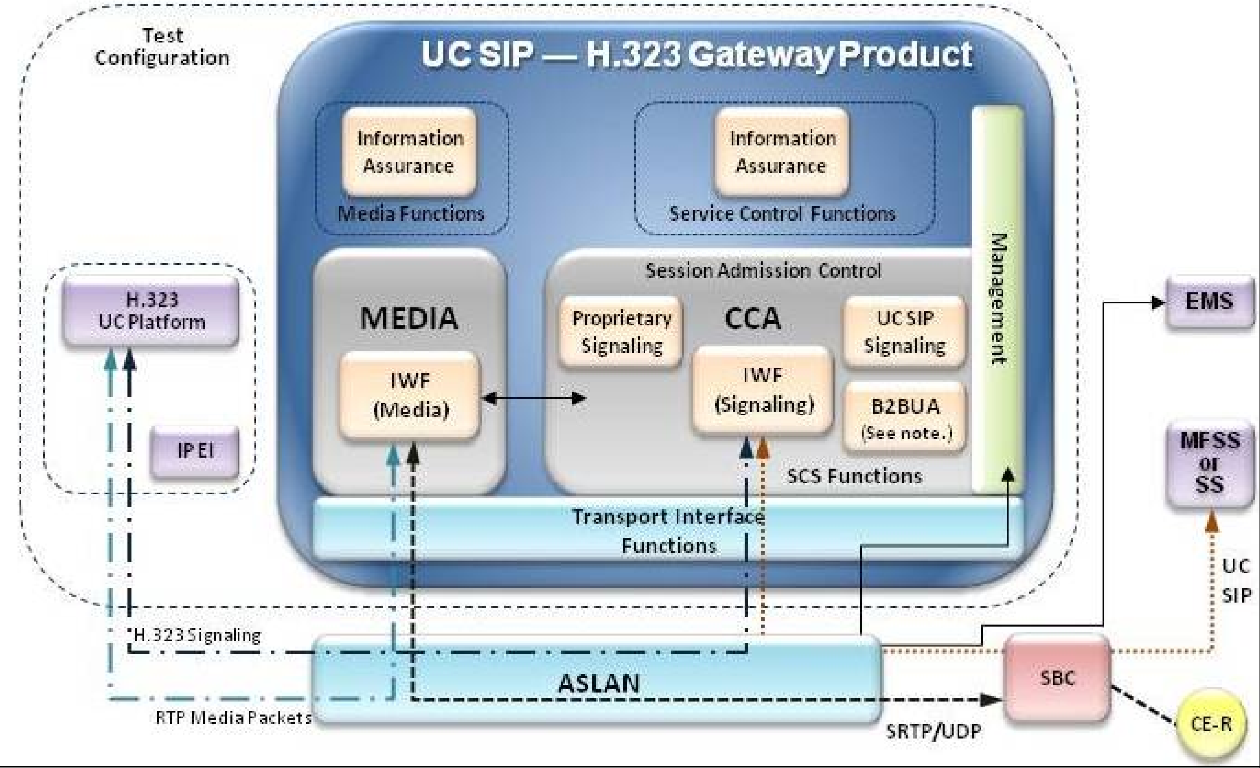
6. OPERATIONAL ARCHITECTURE. Figure 2-1 depicts the Unified Capabilities Requirements (UCR) operational Defense Information Systems Network (DISN) Architecture. Figure 2-2 depicts the reference model for the AS-SIP – H.323 Gateway. The AS-SIP – H.323 Gateway consists of several Session Control and Signaling (SCS) functions performed by the Call Connection Agent (CCA), Inter-working Function (IWF) (for signaling), and IWF (for media). These are connected via H.323 internal interface functions. Connectivity to other networks, long-haul transport systems via an Assured Services Local Area Network (ASLAN), and DISA’s Voice and Video over IP (VVoIP) Network Management System (NMS) are provided by external interfaces.



LEGEND:

DCO	Defense Connection Online	PKI	Public Key Infrastructure
DISA	Defense Information Systems Agency	PSTN	Public Switched Telephone Network
DISN	Defense Information Systems Network	QoS	Quality of Service
DoD	Department of Defense	SBC	Session Border Controller
EI	End Instrument	SC	Session Controller
IAP	Internet Access Point	SS	Softswitch
IM	Instant Messaging	STEP	Standardized Tactical Entry Point
IP	Internet Protocol	UC	Unified Capabilities
ISP	Internet Service Provider	VVoIP	Voice and Video over IP
LAN	Local Area Network	XMPP	Extensible Messaging and Presence Protocol
MCEP	Multi Carrier Entry Point		

Figure 2-2. Operational DISN Architecture



NOTE: B2BUA is applicable when the UC platform signaling protocol is SIP.

LEGEND:

ASLAN	Assured Services Local Area Network	IWF	Interworking Function
AS-SIP	Assured Services Session Initiation Protocol	MFSS	Multifunction Softswitch
B2BUA	Back-to-Back User Agent	RTP	Real-Time Transport Protocol
CCA	Call Connection Agent	SBC	Session Border Controller
CE-R	Customer Edge Router	SCS	Session Control and Signaling
EI	End Instrument	SIP	Session Initiation Protocol
EMS	Element Management System	SRTP	Secure Real-Time Transport Protocol
H.323	Standard for multi-media communications on packet-based networks	SS	Softswitch
		UC	Unified Capabilities
IP	Internet Protocol	UDP	User Datagram Protocol

Figure 2-2. Functional Reference Model – AS-SIP – H.323 Gateway

7. INTEROPERABILITY REQUIREMENTS. The interface, Capability Requirements (CR) and Functional Requirements (FR), Information Assurance (IA), and other requirements for ASLAN infrastructure products are established by Sections 5.3.2.7.5 and 5.4 of Reference (c).

7.1 Interfaces. Table 2-1 depicts the physical AS-SIP to H.323 Gateway interfaces and the associated standards.

Table 2-1. AS-SIP to H.323 Gateway Interface Requirements

Interface	UCR Reference	Criteria (See note 1.)	Applicability
AS-SIP – H.323 Gateway-to-UC WAN Appliance Interface			
10Base-X (See note 2.)	5.3.2.7.5.1	Support minimum threshold CRs/FRs 1-3, 5, 6 and meet interface criteria for IEEE 802.3i.	C
100Base-X (See note 2.)	5.3.2.7.5.1	Support minimum threshold CRs/FRs 1-3, 5, 6 and meet interface criteria for IEEE 802.3.u.	C
1000Base-X (See note 2.)	5.3.2.7.5.1	Support minimum threshold CRs/FRs 1-3, 5, 6 and meet interface criteria for IEEE 802.3z or 802.3ab.	C
AS-SIP – H.323 Gateway-to-NMS Interface			
10/100 Mbps	5.3.2.7.5.3.7	Support minimum threshold CRs/FRs 4, 6 and meet interface criteria for IEEE 802.3i and 802.3u.	R

NOTES:
1. The SUT high-level CR and FR ID numbers depicted in the Threshold CRs/FRs column can be cross-referenced in Table 2-2. These high-level CR/FR requirements refer to a detailed list of requirements provided in Enclosure 3.
2. The UCR 2008, Change 3, does not specify minimum interfaces for an AS-SIP – ITU-T H.323 Gateway.

LEGEND:

802.3ab	1000BaseT Gbps Ethernet over twisted pair at 1 Gbps (125 Mbps)	IEEE	Institute of Electrical and Electronics Engineers
802.3i	10BaseT Mbps over twisted pair	ITU-T	International Telecommunication Union - Telecommunication Standardization Sector
802.3u	Standard for carrier sense multiple access with collision detection at 100 Mbps	H.323	ITU-T Standard for multi-media communications on packet-based networks
802.3z	Gigabit Ethernet Standard	Mbps	Megabits per second
AS-SIP	Assured Services Session Initiation Protocol	NMS	Network Management System
C	Conditional	R	Required
CR	Capability Requirement	SUT	System Under Test
FR	Functional Requirement	UC	Unified Capabilities
Gbps	Gigabits per second	UCR	Unified Capabilities Requirements
ID	Identification	WAN	Wide Area Network

7.2 CR and FR. AS-SIP to H.323 Gateway products have required and conditional features and capabilities that are established by Section 5.3.2.7.5 of the UCR. The SUT does not need to provide non-critical (conditional) requirements. If they are provided, they must function according to the specified requirements. The SUT requirements are listed in Table 2-2. Detailed CR/FR requirements are provided in Table 3-1 of Enclosure 3.

Table 2-2. AS-SIP to ITU-T H.323 Gateway CRs and FRs

CR/FR ID	Capability/Function	Applicability (See note.)	UCR Reference
1	**AS-SIP – H.323 Gateway General Requirements**		
	Call request from H.323 UC signaling platform	Required	5.3.2.7.5.1.2
	Initial routine AS-SIP invite from the WAN SS/MFSS	Required	5.3.2.7.5.1.3
	Process initial precedence AS-SIP invites from the WAN SS/MFSS via the EBC	Required	5.3.2.7.5.1.4
	AS-SIP – H.323 Gateway support for VoIP and Video Signaling Interfaces	Required	5.3.2.7.5.3.1
	CCA requirements.	Required	5.3.2.7.5.3.1.1
	Role of IWF within the CCA	Required	5.3.2.7.5.3.1.1.1
	Resource priority header	Required	5.3.2.7.5.3.1.2
	Mapping of Telephony Number into SIP URI	Required	5.3.2.7.5.3.1.3
	Support for audio and video codecs	Required	5.3.2.7.5.3.2
	Product Quality Factors	Required	5.3.2.7.5.3.8
2	**SAC**		
	The AS-SIP – H.323 Gateway must implement call count thresholds for voice sessions and for video sessions in order to perform Session Admission Control (SAC)	Required	5.3.2.7.5.1.1
	SAC	Required	5.3.2.7.5.3.1.4.1
	Directionalization	Conditional	5.3.2.7.5.3.1.4.2
	Code Blocking	Required	5.3.2.7.5.3.1.4.3
	Configuration of total voice call thresholds and total video call thresholds	Required	5.3.2.7.5.3.1.4.4
	Configuration of outbound voice call thresholds, inbound voice call thresholds, outbound video call thresholds, and inbound video call thresholds.	Required	5.3.2.7.5.3.1.4.5
	SAC enforcement	Required	5.3.2.7.5.3.1.4.6 5.3.2.7.5.3.1.4.7 5.3.2.7.5.3.1.4.8
3	**AS-SIP – H.323 Gateway Management Function**		
	Section 5.3.2.17.1, Voice and Video Network Management Domain	Required	5.3.2.7.5.3.5
	Section 5.3.2.17.2, General Management Requirements	Required	5.3.2.7.5.3.5
	The AS-SIP – H.323 Gateway must support one pair of Ethernet management interfaces where one management interface is for communication with a local EMS and one management interface is for communication with a remote EMS. In addition, the AS-SIP – H.323 Gateway must support at least one additional Ethernet interface for carrying signaling and media streams for VVoIP traffic.	Required	5.3.2.7.5.3.5
	Section 5.3.2.17.3.1, Fault Management	Required	5.3.2.7.5.3.5
	Section 5.3.2.17.3.2.1, Read-Write Access to CM Data by the RTS EMS	Required	5.3.2.7.5.3.5
	Section 5.3.2.17.3.4.1, Near-Real-Time Network Performance Monitoring	Required	5.3.2.7.5.3.5
	Section 5.3.2.17.3.4.2, Remote Network Management Commands (the LSC requirements apply to the AS-SIP – H.323 Gateway with the exception of Section 5.3.2.17.3.4.2.14, PEI/GEI Origination Capability Control)	Required	5.3.2.7.5.3.5
	Section 5.3.2.17.3.5, Security Management	Required	5.3.2.7.5.3.5
	Section 5.3.2.17.4, Data Classification	Required	5.3.2.7.5.3.5
	Section 5.3.2.17.5, Management of Appliance Software	Required	5.3.2.7.5.3.5
	Requirement 5.3.2.18.1, NM Requirements for CE Routers and EBCs	Required	5.3.2.7.5.3.5
	Section 5.3.2.18.2, Management Requirements for the ASAC (use these requirements for SAC only)	Required	5.3.2.7.5.3.5
	Section 5.3.2.18.3.1.1, CCA Support for Capacity Installation, but not including Section 5.3.2.18.3.1.1.1, MG-Related Configuration, and Section 5.3.2.18.3.1.1.2, SG-Related Data)	Required	5.3.2.7.5.3.5
	Section 5.3.2.18.3.3, CCA Support for Fault Localization	Required	5.3.2.7.5.3.5
	Section 5.3.2.18.3.4, CCA Support for Testing	Required	5.3.2.7.5.3.5

Table 2-2. AS-SIP to ITU-T H.323 Gateway CRs and FRs (continued)

CR/FR ID	Capability/Function	Applicability (See note.)	UCR Reference
4	**Interface to the DISA NMS**		
	The AS-SIP – H.323 Gateway must provide a 10/100 Mbps interface to the DISA NMS IAW section 5.3.2.4.4	Required	5.3.2.7.5.3.7
5	**IPv6 Requirements**		
	Product Requirements for NA/SS	Required	5.3.5.4
6	**Information Assurance**		
	The AS-SIP – H.323 Gateway must meet IA requirements IAW UCR 2008, Change 3, Section 5.4 for a media gateway.	Required	5.3.2.7.5.3.3

NOTE: The annotation of 'required' refers to a high-level requirement category. The applicability of each sub-requirement is provided in Enclosure 3. The SUT does not need to provide conditional requirements. However, if a capability is provided, it must function according to the specified requirements.

LEGEND:

ASAC	Assured Services Admission Control
AS-SIP	Assured Services Session Initiation Protocol
CCA	Call Connection Agent
CE	Customer Edge
CM	Configuration Management
CR	Capability Requirement
DISA	Defense Information Systems Agency
EBC	Edge Boundary Controller
EMS	Element Management System
FR	Functional Requirement
GEI	Generic End Instrument
IA	Information Assurance
IAW	in accordance with
ID	Identification
IPv6	Internet Protocol version 6
ITU-T	International Telecommunication Union - Telecommunication Standardization Sector
H.323	Standard for multi-media communications on packet-based networks
LoC	Letters of Compliance
LSC	Local Session Controller
Mbps	Megabits per second
MFSS	Multi-function Softswitch
NA/SS	Network Appliance/Simple Server
NM	Network Management
NMS	Network Management System
PEI	Proprietary End Instrument
RTS	Real Time Services
SAC	Session Admission Control
SIP	Session Initiation Protocol
SS	Softswitch
SUT	System Under Test
UC	Unified Capabilities
UCR	Unified Capabilities Requirements
URI	Uniform Resource Indicator
VoIP	Voice over Internet Protocol
VVoIP	Voice and Video over Internet Protocol
WAN	Wide Area Network

7.3 Information Assurance (IA). Table 2-3 details the IA requirements applicable to the AS-SIP to H.323 Gateway products.

Table 2-3. AS-SIP to H.323 Gateway IA Requirements

Requirement	Applicability (See note.)	UCR Reference	Criteria
General Requirements	Required	5.4.6.2	Detailed requirements and associated criteria for gateways are listed in Reference (c), Section 5.4.
Authentication	Required	5.4.6.2.1	
Integrity	Required	5.4.6.2.2	
Confidentiality	Required	5.4.6.2.3	
Non-Repudiation	Required	5.4.6.2.4	
Availability	Required	5.4.6.2.5	

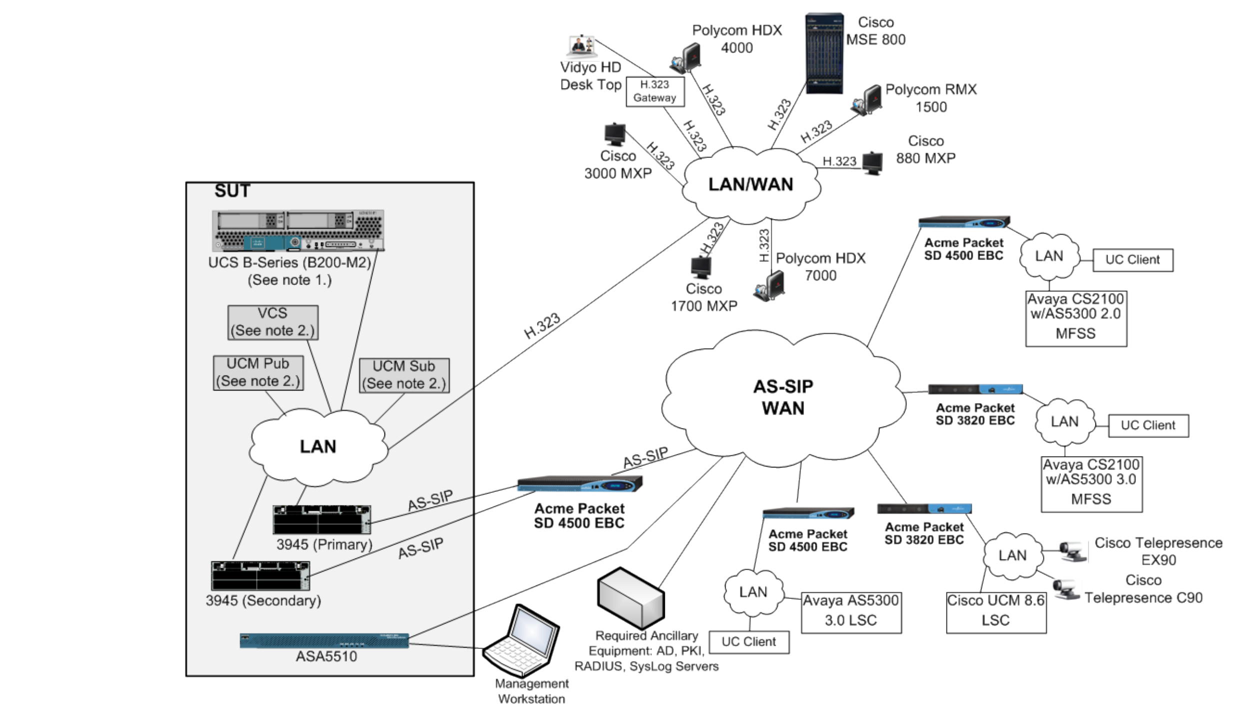
NOTE: The annotation of 'required' refers to a high-level requirement category. Refers to IA requirements for UCR 2008, Change 3, Section 5.4.

Table 2-3. AS-SIP to H.323 Gateway IA Requirements

LEGEND:			
AS-SIP	Assured Services Session Initiation Protocol	ITU-T	International Telecommunications Union – Telecommunication Standardization Sector
IA	Information Assurance	UCR	Unified Capabilities Requirements

7.4 Other. None

8. TEST NETWORK DESCRIPTION. The SUT was tested at JITC in a manner and configuration similar to that of a notional operational environment. Testing the system's required functions and features was conducted using the configuration depicted in Figure 2-3.



NOTES:
1. Supported hardware configurations can be found by selecting the "Cisco Unified Communications on the Cisco Unified Computing System" link at the following URL: www.cisco.com/go/swonly.
2. The VCS, UCM Publisher, and UCM Subscriber were tested as Virtual Machines hosted on the UCS B200 M2 server. The VCS, UCM Publisher, and UCM Subscriber are certified as VMs on the UC server or as separate components.

LEGEND:

AS-SIP	Assured Services Session Initiation Protocol	UC	Unified Capabilities
EBC	Edge Boundary Controller	UCM	Unified Communications Manager
LAN	Local Area Network	UCS	Unified Computing System
LSC	Local Session Controller	VCS	Video Communications Server
MFSS	Multifunction Softswitch	VM	Virtual Machine
Pub	Publisher	WAN	Wide Area Network
Sub	Subscriber		

Figure 2-3. SUT Test Configuration

9. SYSTEM CONFIGURATIONS. Table 2-4 provides the system configurations and hardware and software components tested with the SUT. The SUT was tested in an operationally realistic environment to determine its interoperability capability with associated network devices and network traffic.

Table 2-4. Tested System Configurations

Equipment	Software
Avaya CS2100 with AS5300 (MFSS)	CS2100 Release SE09.1 with AS5300 Release 2.0
Avaya CS2100 with AS5300 (MFSS)	CS2100 Release SE09.1 with AS5300 Release 3.0
Acme Packet 3820 (EBC)	SCX 6.3.0 Maintenance Release (MR)2 Patch (P)5 (Build 429)
Acme Packet 4500 (EBC)	SCX 6.3.0 Maintenance Release (MR)2 Patch (P)5 (Build 429)
Avaya AS5300 (LSC)	3.0
Cisco Unified Communications Manager (LSC)	8.6.1
Avaya UC Client (softphone)	UC Client Version: 8.1.5126
Cisco C90 (Cisco VTC codec)	TC 6.1.0
Cisco EX90 (Cisco VTC codec)	TC 6.1.0
Cisco Media Service Engine (MSE) 8000	2.0 (1.13)
Vidyo (VTC)	2.1.2.56_D
Required Ancillary Equipment	Active Directory
	Public Key Infrastructure
	Remote Authentication Dial-In User Service
	SysLog Server
	Site-provided Management Workstation Microsoft Windows 7
SUT (See note 1.)	**Release**
Unified Communications Server (**B-Series5108**) with **B200M2** with ESXi 5.1 **(See note 2.)**	8.6.1 20011-4
UCM Publisher (See note 3.)	8.6.1 20011-4
UCM Subscriber (See note 3.)	8.6.1 20011-4
Interworking Gateway running on an ISRG2 (**3945**, 3925, 3945E, 3925E)	15.2(4)M3
TelePresence Video Communications Server (See note 3.)	X8.1.1
Adaptive Security Appliance **5510**, 5505, 5520, 5540, 5585-SSP10, 5585-SSP20, 5585-SSP30, 5585-SSP40	8.4(3)

NOTES:
1. Components bolded and underlined were tested by JITC. The other components in the family series were not tested; however, they utilize the same software and similar hardware and JITC analysis determined them to be functionally identical for interoperability certification purposes and they are also certified for joint use.
2. Supported hardware configurations can be found by selecting the "Cisco Unified Communications on the Cisco Unified Computing System" link at the following URL: www.cisco.com/go/swonly.
3. The VCS, UCM Publisher, and UCM Subscriber were tested as Virtual Machines hosted on the UCS B200 M2 server. The VCS, UCM Publisher, and UCM Subscriber are certified as VMs on the UC server or as separate components.

LEGEND:

EBC	Edge Boundary Controller	SUT	System Under Test
ISRG2	Integrated Service Router Generation 2	UC	Unified Communications
JITC	Joint Interoperability Test Command	UCM	Unified Communications Manager
LSC	Local Session Controller	VCS	Video Communications Server
MFSS	Multifunction Softswitch	VTC	Video Telecommunications

10. TESTING LIMITATIONS. None.

11. INTEROPERABILITY EVALUATION RESULTS. The SUT meets the critical interoperability requirements for an AS-SIP to H.323 Gateway in accordance with UCR 2008, Change 3, Section 5.3.2.7.5, and is certified for joint use with other network infrastructure products listed on the UC Approved Products List (APL). Additional discussion regarding specific testing results is located in subsequent paragraphs. The

AS-SIP – H.323 Gateway is a VVoIP interworking appliance, and its purpose is to enable the interconnection and interoperation of H.323 IP-based UC signaling platforms and their associated IP EIs with the DISN UC system to support End-to-End (E2E) voice and video sessions. The Government has adopted RFC 4123 – Session Initiation Protocol (SIP) – H.323 Interworking Requirements as the document which describes the requirements for the AS-SIP – H.323 Gateway. Internet draft-agrawal-sip-h323-interworking-01.txt is cited as guidance to be used in implementing the AS-SIP – H.323 Gateway. The requirements in this section are additional Government requirements. The AS-SIP – H.323 Gateway SUT is a standalone SUT for testing purposes.

11.1 Interfaces. The interface status of the SUT is provided in Table 2-5. From a signaling perspective, the AS-SIP – H.323 Gateway must offer an AS-SIP-compliant signaling interface that provides end-to-end signaling interoperability between the AS-SIP – H.323 Gateway SUT and the AS-SIP signaling appliances of the DISN UC Wide Area Network (WAN) system. From a media perspective, the AS-SIP – H.323 Gateway must offer a UCR-compliant bearer interface that provides E2E interoperability for voice and video media packets between the ASSIP – H.323 Gateway SUT and EBCs, IP EIs of LSCs, MGs, and AS-SIP EIs. The AS-SIP – H.323 Gateway must interwork the voice and video media packets generated by the IP EIs served by the IP-based UC signaling platform and intended for a destination outside the H.323 system enclave to UCR-compliant Secure Real-Time Transport Protocol (SRTP)/User Datagram Protocol (UDP) packets having the appropriate Differentiated Services Code Point (DSCP). Similarly, UCR-compliant SRTP/UDP voice and video media packets received from the UC WAN and intended for the IP EIs served by the H.323 UC signaling platform must be interworked by the AS-SIP – H.323 Gateway into the H.323 media packets supported by the IP EIs. The UCR does not mandate the connectivity, interface, or protocol requirements within the AS-SIP – H.323 Gateway. However, the SUT must meet the minimum requirements for connectivity to an EBC. The SUT met the interface requirements with 10/100/1000 Megabits per second (Mbps) interfaces through both testing and vendor's Letters of Compliance (LoC).

Table 2-5. SUT Interface Requirements Status

Interface	Applicability	UCR 2008, Change 3	Threshold CR/FR (See note 1.)	Status	Remarks
AS-SIP – H.323 Gateway-to-UC WAN Appliance Interface					
10Base-X	C (See note 2.)	5.3.2.7.5.1	1-3, 5, 6	Met	The SUT met the critical CRs and FRs with the following IEEE standard: 802.3i (10BaseT).
100Base-X	C (See note 2.)	5.3.2.7.5.1	1-3, 5, 6	Met	The SUT met the critical CRs and FRs with the following IEEE standard: 802.3u (100BaseT).
1000Base-X	C (See note 2.)	5.3.2.7.5.1	1-3, 5, 6	Met	The SUT met CR and FRs with the following IEEE standard: 802.3ab (1000BaseT).
AS-SIP – H.323 Gateway-to-NMS Interface					
10/100 Mbps	R	5.3.2.7.5.3.7	4, 6	Partially Met	See note 3.

Table 2-5. SUT Interface Interoperability Status (continued)

NOTES:
1. The SUT high-level CR and FR ID numbers depicted in the Threshold CRs/FRs column can be cross-referenced in Table 3. These high-level CR/FR requirements refer to a detailed list of requirements provided in Enclosure 3.
2. The UCR 2008, Change 3, does not specify minimum interfaces for an AS-SIP –H.323 Gateway.
3. The UCR 2008 Change 3, section 5.3.2.7.5.3.7, states that the AS-SIP – H.323 Gateway must provide a 10/100-Mbps Ethernet interface to the DISA NMS, separate from the local management interface. The SUT supports only one physical interface. DISA adjudicated this as minor with the condition of fielding to be included in the vendor's deployment guide stipulating that local NM access and remote NM access will be provided via a single interface.

LEGEND:

802.3ab	1000BaseT Gbps Ethernet over twisted pair at 1 Gbps (125 Mbps)
802.3i	10BaseT Mbps over twisted pair
802.3u	Standard for carrier sense multiple access with collision detection at 100 Mbps
AS-SIP	Assured Services Session Initiation Protocol
C	Conditional
CR	Capability Requirement
DISA	Defense Information Systems Agency
FR	Functional Requirement
Gbps	Gigabits per second
ID	Identification
IEEE	Institute of Electrical and Electronics Engineers
H.323	Standard for multi-media communications on packet-based networks
Mbps	Megabits per second
NM	Network Management
NMS	Network Management System
R	Required
SUT	System Under Test
UC	Unified Capabilities
UCR	Unified Capabilities Requirements
WAN	Wide Area Network

11.2 CR and FR. The SUT CR and FR status is depicted in Table 2-6. Detailed CR/FR requirements are provided in Enclosure 3, Table 3-1.

Table 2-6. SUT CR and FR Status

CR/FR ID	Capability/Function	Applicability (See note 1.)	UCR Reference	Status
1	**AS-SIP – H.323 Gateway General Requirements**			
	Call request from H.323 UC signaling platform	Required	5.3.2.7.5.1.2	Met
	Initial routine AS-SIP invite from the WAN SS/MFSS	Required	5.3.2.7.5.1.3	Met
	Process initial precedence AS-SIP invites from the WAN SS/MFSS via the EBC	Required	5.3.2.7.5.1.4	Met (See note 2.)
	AS-SIP – H.323 Gateway support for VoIP and Video Signaling Interfaces	Required	5.3.2.7.5.3.1	Met
	CCA requirements.	Required	5.3.2.7.5.3.1.1	Met
	Role of IWF within the CCA	Required	5.3.2.7.5.3.1.1.1	Met
	Resource priority header	Required	5.3.2.7.5.3.1.2	Met
	Mapping of Telephony Number into SIP URI	Required	5.3.2.7.5.3.1.3	Met
	Support for audio and video codecs	Required	5.3.2.7.5.3.2	Met
	Product Quality Factors	Required	5.3.2.7.5.3.8	Met (See note 3.)

Table 2-6. SUT CR and FR Status (continued)

CR/FR ID	Capability/Function	Applicability (See note 1.)	UCR Reference	Status
2	**SAC**			
	The AS-SIP – H.323 Gateway must implement call count thresholds for voice sessions and for video sessions in order to perform SAC.	Required	5.3.2.7.5.1.1	Met
	SAC	Required	5.3.2.7.5.3.1.4.1	Met
	Directionalization	Conditional	5.3.2.7.5.3.1.4.2	Not Supported
	Code Blocking	Required	5.3.2.7.5.3.1.4.3	Met
	Configuration of total voice call thresholds and total video call thresholds	Required	5.3.2.7.5.3.1.4.4	Met
	Configuration of outbound voice call thresholds, inbound voice call thresholds, outbound video call thresholds, and inbound video call thresholds.	Conditional	5.3.2.7.5.3.1.4.5	Not Supported
	SAC enforcement	Required	5.3.2.7.5.3.1.4.6 5.3.2.7.5.3.1.4.7 5.3.2.7.5.3.1.4.8	Met
3	**AS-SIP – H.323 Gateway Management Function**			
	Section 5.3.2.17.1, Voice and Video Network Management Domain	Required	5.3.2.7.5.3.5	Met
	Section 5.3.2.17.2, General Management Requirements	Required	5.3.2.7.5.3.5	Partially Met (See note 4.)
	The AS-SIP – H.323 Gateway must support one pair of Ethernet management interfaces where one management interface is for communication with a local EMS and one management interface is for communication with a remote EMS. In addition, the AS-SIP – H.323 Gateway must support at least one additional Ethernet interface for carrying signaling and media streams for VVoIP traffic.	Required	5.3.2.7.5.3.5	Not Met (See note 5.)
	Section 5.3.2.17.3.1, Fault Management	Required	5.3.2.7.5.3.5	Partially Met (See note 6.)
	Section 5.3.2.17.3.2.1, Read-Write Access to CM Data by the RTS EMS	Required	5.3.2.7.5.3.5	Met
	Section 5.3.2.17.3.4.1, Near-Real-Time Network Performance Monitoring	Required	5.3.2.7.5.3.5	Partially Met (See note 7.)
	Section 5.3.2.17.3.4.2, Remote Network Management Commands (the LSC requirements apply to the AS-SIP – H.323 Gateway with the exception of Section 5.3.2.17.3.4.2.14, PEI/GEI Origination Capability Control)	Required	5.3.2.7.5.3.5	Met
	Section 5.3.2.17.3.5, Security Management	Required	5.3.2.7.5.3.5	Met
	Requirement 5.3.2.18.1, NM Requirements for CE Routers and EBCs	Required	5.3.2.7.5.3.5	Not Met (See note 8.)
	Section 5.3.2.18.2, Management Requirements for the ASAC (use these requirements for SAC only)	Required	5.3.2.7.5.3.5	Partially Met (See note 9.)
4	**Interface to the DISA NMS**			
	The AS-SIP – H.323 Gateway must provide a 10/100 Mbps interface to the DISA NMS IAW section 5.3.2.4.4	Required	5.3.2.7.5.3.7	Not Met (See note 5.)
5	**IPv6**			
	Product Requirements for NA/SS	Required	5.3.5.4	Partially Met (See notes 10, 11, 12, 13.)
6	**Information Assurance**			
	The AS-SIP – H.323 Gateway must meet IA requirements IAW UCR 2008, Change 3, Section 5.4 for a media gateway.	Required	5.3.2.7.5.3.3	Met (See note 14.)

Table 2-6. SUT CR and FR Status (continued)

NOTES:
1. The annotation of 'required' refers to a high-level requirement category. The applicability of each sub-requirement is provided in Enclosure 3. The system under test does not need to provide conditional requirements. However, if a capability is provided, it must function according to the specified requirements.
2. The SUT meets the requirement by diverting the INVITE to the attendant when calls above ROUTINE are placed from the AS-SIP enclave to the H.323 non Command and Control (C2) enclave.
3. The UCR states the AS-SIP to H.323 Gateway can meet either Medium Availability or High Availability. High Availability is supported for the AS-SIP interface; however, the H.323 interface does not meet the High Availability product quality factors. Therefore, the SUT is certified as Medium Availability.
4. The SUT partially complies with the general management requirements. The SUT meets the data reporting requirements; however, it does not use SNMP or eXtensible Markup Language (XML) to do so. DISA adjudicated this as minor with the condition of fielding to be included in the vendor's deployment guide stipulating an alternative procedure to manage the gateway.
5. The SUT does not provide an NM interface to the DISA NMS. The SUT supports only one physical interface. DISA adjudicated this as minor with the condition of fielding to be included in the vendor's deployment guide stipulating that local NM access and remote NM access will be provided via a single interface.
6. The SUT partially complies with the fault management requirements. Fault reporting is not performed via SNMP. DISA adjudicated this as minor with the condition of fielding to be included in the vendor's deployment guide stipulating an alternative procedure to manage the gateway.
7. The SUT partially complies with performance monitoring capability per the reference. This capability is available using (other) existing UCM tools. Cisco is unable to verify SUT capabilities for EMS interoperability. DISA adjudicated this as minor with the condition of fielding to be included in the vendor's deployment guide stipulating an alternative procedure to manage the gateway.
8. The SUT must comply with the following NM requirements: Faults will be reported IAW RFCs 1215 and 3418. Standard CM and PM information shall be presented IAW RFCs 1213 and 3418. DISA adjudicated this as minor with the condition of fielding to be included in the vendor's deployment guide stipulating an alternative procedure to manage the gateway.
9. The UCR requirement states that the SUT must permit reading of the VoIP, video, and other session counts (as applicable) from the VVoIP EMS. Call counts can be read from SUT management application; however, the vendor does not know if they can be read by VVoIP EMS. DISA adjudicated this as minor with the condition of fielding to be included in the vendor's deployment guide stipulating an alternative procedure to manage the gateway.
10. The SUT meets the requirements for IPv4/IPv6 dual stack with the following minor exception. The SUT does not support dual stack for H.323. DISA adjudicated this as minor. There is no mature standard for dual stack with H.323 protocol. The SUT is certified for IPv4 only on the H.323 side of the gateway. The SUT will be fielded with IPv4 on the H.323 side of the gateway.
11. The SUT VCS does not comply with Multicast Listener Discovery (MLD) as described in RFC 2710. DISA adjudicated this as minor. The SUT will be fielded with IPv4 on the H.323 side of the gateway.
12. The SUT partially complies with assigning the DSCP tagging requirements. The SUT VCS can be assigned any DSCP value 0-63, however this value is assigned to all types of packets to include OA&M, Signaling and Media. DISA accepted the vendor's PoA&M and adjudicated this as minor.
13. The SUT supports IPv6 Neighbor Discovery and SLAAC in accordance with RFCs 2461 and 2462 and not RFCs 4861 and 4862 as required by the UCR. During testing, there were no interoperability discrepancies related to IPv6 Neighbor Discovery or SLAAC. DISA accepted the vendor's PoA&M and adjudicated this as minor.
14. Security is tested by DISA-led Information Assurance test teams and the results published in a separate report, Reference (e).

Table 2-6. SUT CR and FR Status (continued)

LEGEND:

ASAC	Assured Services Admission Control	NM	Network Management
AS-SIP	Assured Services Session Initiation Protocol	NMS	Network Management System
CCA	Call Connection Agent	OA&M	Operations, Administration, and Maintenance
CE	Customer Edge	PEI	Proprietary End Instrument
CM	Configuration Management	PM	Performance Management
CR	Capability Requirement	PoA&M	Plan of Action and Milestones
DISA	Defense Information Systems Agency	RFC	Request for Comments
DSCP	Differentiated Services Code Point	RTS	Real Time Services
EBC	Edge Boundary Controller	SAC	Session Admission Control
EMS	Element Management System	SIP	Session Initiation Protocol
FR	Functional Requirement	SLAAC	Stateless Auto Address Configuration
GEI	Generic End Instrument	SNMP	Simple Network Management Protocol
IA	Information Assurance	SS	Softswitch
IAW	in accordance with	SUT	System Under Test
ID	Identification	UC	Unified Capabilities
IPv4	Internet Protocol version 4	UCM	Unified Communications Manager
IPv6	Internet Protocol version 6	UCR	Unified Capabilities Requirements
IWF	Inter-working Function	URI	Uniform Resource Indicator
H.323	Standard for multi-media communications on packet-based networks	VCS	Video Communication Server
Mbps	Megabits per second	VoIP	Voice over Internet Protocol
MFSS	Multi-function Softswitch	VVoIP	Voice and Video over Internet Protocol
NA/SS	Network Appliance/Simple Server	WAN	Wide Area Network

a. AS-SIP – H.323 Gateway General Requirements.

(1) The UCR 2008 Change 3, section 5.3.2.7.5.1.2, states that when the AS-SIP – H.323 Gateway receives a call request from the H.323 UC signaling platform then the AS-SIP – H.323 Gateway must meet the requirements in the subparagraphs below.

(a) Check the appropriate (voice or video) call count (and outbound call count in the case of directionalization) to determine whether there are available bandwidth resources to support the call request. The SUT met this requirement through testing.

(b) If the new call request would not exceed the appropriate (voice or video) call count threshold (and outbound call count threshold) then the AS-SIP – H.323 Gateway interworks the call request in accordance with the requirements in the subparagraphs below. The SUT met this requirement through testing.

1. Incrementing the call count (and outbound call count in the case of directionalization).

2. Generating a “routine” level AS-SIP INVITE that advertises equivalent capabilities to those specified in the received call request.

3. Adding a CCA-ID parameter to the Contact header.

4. Adding a route set comprising two Route headers where the first Route header is the SIP Uniform Resource Indicator (URI) for the EBC at the enclave, and the second Route header is the SIP URI for the EBC serving the WAN SS/MFSS.

5. Forwarding the INVITE message to the EBC at the enclave.

(c) If the appropriate (voice or video) call count (or outbound call count) is at threshold or the call request would cause the AS-SIP – H.323 Gateway to exceed the call count threshold (or outbound call count threshold) then the AS-SIP – H.323 Gateway must reject the call. The SUT met this requirement through testing.

(2) The UCR 2008 Change 3, section 5.3.2.7.5.1.3, states that when an AS-SIP – H.323 Gateway receives an initial routine AS-SIP INVITE (i.e., not a re-INVITE) from the WAN SS/MFSS (via the EBC), then the AS-SIP – H.323 Gateway must meet the requirements in the subparagraphs below.

(a) Check the appropriate (voice or video) call count (and inbound call count in the case of directionalization) to determine whether there are available bandwidth resources to support the call request. The SUT met this requirement through testing.

(b) If the new call request would not exceed the appropriate (voice or video) call count threshold (and inbound call count threshold) then the AS-SIP – H.323 Gateway increments the appropriate (voice or video) call count (and inbound call count) and interworks the INVITE to H.323. The SUT met this requirement through testing.

(c) If the appropriate (voice or video) call count (or inbound call count) is at threshold or the call request would cause the AS-SIP – H.323 Gateway to exceed the appropriate (voice or video) call count threshold (or inbound call count threshold) then the ASSIP – H.323 Gateway must reject the call. The response message is 488 (Not Acceptable Here) and should include a Warning header field with warning code 370 (Insufficient Bandwidth). The SUT met this requirement through testing and the vendor’s LoC.

(3) The UCR 2008 Change 3, section 5.3.2.7.5.1.4, states that the AS-SIP – H.323 Gateway must support one of the following two methods for processing initial precedence AS-SIP INVITEs received from the WAN SS/MFSS via the EBC and the choice of method must be software configurable.

(a) Upon receipt of the initial precedence AS-SIP INVITE request the AS-SIP – H.323 Gateway diverts the precedence INVITE to the attendant. The SUT met this requirement through testing. The SUT diverts the INVITE to the attendant when calls above ROUTINE are placed from the AS-SIP enclave to the H.323 non Command and Control (C2) enclave.

(b) Upon receipt of the initial precedence AS-SIP INVITE request, the AS-SIP – H.323 Gateway determines whether the appropriate (voice or video) call count (or inbound call count in the case of directionalization) is at threshold or whether the call request would cause the AS-SIP – H.323 Gateway to exceed the appropriate (voice or video) call count threshold or inbound call count threshold:

1. If the precedence AS-SIP INVITE would cause the appropriate (voice or video) call count threshold (or inbound call count threshold) to be exceeded, then the precedence AS-SIP INVITE is forwarded to the attendant. The AS-SIP – H.323 Gateway must not conduct preemption on behalf of an inbound precedence AS-SIP INVITE.

2. If the precedence AS-SIP INVITE would not cause the appropriate call count threshold (or inbound call count threshold) to be exceeded, then the ASSIP – H.323 Gateway treats the inbound precedence AS-SIP INVITE request as if it were a routine inbound call request and increments the appropriate (voice or video) call count (and inbound call count) and interworks the INVITE to platform. The SUT diverts all precedence above ROUTINE calls from the UC AS-SIP network to the attendant, which meets this requirement. The SUT met this requirement through testing.

(4) The UCR 2008 Change 3, section 5.3.2.7.5.3.1, states that the AS-SIP – ITU-T H.323 Gateway support VoIP and Video Signaling Interfaces in accordance with (IAW) Table 5.3.2.7-2. The SUT met this requirement through testing.

(5) The UCR 2008 Change 3, section 5.3.2.7.5.3.1.1, states that the CCA is part of the SCS functions and includes the requirements in the subparagraphs below. The SUT met these requirements through testing.

(a) AS-SIP signaling protocol implementation for voice and video calls.

(b) H.323 signaling protocol implementation for voice and video calls (where signaling protocol implementation refers to the signaling being used by the H.323 UC signaling platform).

(c) Control of sessions within the AS-SIP – H.323 Gateway including H.323 sessions between the AS-SIP – H.323 Gateway and the H.323 UC signaling platform and AS-SIP sessions between the AS-SIP – H.323 Gateway and the serving WAN SS/MFSS.

(d) Support for interactions with other network appliance functions including Admission control, Information Assurance, Media interworking, and Appliance Management functions.

(6) The UCR 2008 Change 3, section 5.3.2.7.5.3.1.1, states that the role of the IWF within the CCA is to interwork the messages of the H.323 VoIP signaling protocol into AS-SIP signaling messages and interwork AS-SIP signaling messages into

messages of the H.323 VoIP signaling protocol. The SUT met this requirement through testing.

(7) The UCR 2008 Change 3, section 5.3.2.7.5.3.1.2, states that the AS-SIP – H.323 Gateway does not conduct preemption. Whenever the AS-SIP – H.323 Gateway receives a H.323 signaling message from the H.323 UC signaling platform that translates it into an INVITE, UPDATE, or REFER request, then the AS-SIP – H.323 Gateway must generate a Resource-Priority header having a ROUTINE priority level IAW Section 5.3.4.10.2. Whenever the AS-SIP – H.323 Gateway receives an INVITE, UPDATE, or REFER request from the WAN SS/MFSS via the EBC, then the AS-SIP – H.323 Gateway must process the Resource-Priority header to distinguish a ROUTINE call from a precedence call. The SUT met this requirement through testing.

(8) The UCR 2008 Change 3, section 5.3.2.7.5.3.1.3, states that when the AS-SIP – H.323 Gateway receives a call request from the H.323 UC signaling platform, then the AS-SIP – H.323 Gateway must map the telephony numbers received from the initial H.323 signaling message to SIP URIs IAW Section 5.3.4.14.3, SIP URI and Mapping of Telephony Number into SIP URI, and UCR Section 5.3.4.7.6, SIP URI and Mapping of Telephone Number into SIP URI. The SUT met this requirement through testing and the vendor's LoC.

(9) The UCR 2008 Change 3, section 5.3.2.7.5.3.2, states that the AS-SIP – H.323 Gateway must support the audio codecs in Section 5.3.2.6.1.2. The gateway must comply with Section 5.3.2.6.1.4. The gateway must support the audio and video codes as specified in Section 5.3.2.6.2.2. The SUT met this requirement through testing and the vendor's LoC.

(10) The UCR 2008 Change 3, section 5.3.2.7.5.3.8, states that the AS-SIP – H.323 Gateway shall meet the product quality factors specified in Section 5.3.2.5.2. The SUT met this requirement through testing and the vendor's LoC. The UCR states the AS-SIP to H.323 Gateway can meet either Medium Availability or High Availability. High Availability is supported for the AS-SIP interface; however, the H.323 interface does not meet the High Availability product quality factors. Therefore, the SUT is certified as Medium Availability.

b. Session Admission Control (SAC).

(1) The UCR 2008 Change 3, section 5.3.2.7.5.3.1.4.1, states that the AS-SIP – H.323 Gateway must conduct SAC as detailed in this section in place of the ASAC required of LSCs. The SUT met this requirement through testing and the vendor's LoC.

(2) The UCR 2008 Change 3, section 5.3.2.7.5.3.1.4.2, states that the AS-SIP – H.323 Gateway must support directionalization. However, the UCR 2008, Change 3, section 5.3.2.7.5, states that directionalization is conditional for an AS-SIP – H.323 gateway. The SUT does not support this conditional requirement.

(3) The UCR 2008 Change 3, section 5.3.2.7.5.3.1.4.3, states that the AS-SIP – H.323 Gateway must support code blocking. The SUT met this requirement through testing and the vendor's LoC.

(4) The UCR 2008 Change 3, section 5.3.2.7.5.3.1.4.4, states that the AS-SIP – H.323 Gateway must support configuration of total voice call thresholds and total video call thresholds. The SUT met this requirement through testing.

(5) The UCR 2008 Change 3, section 5.3.2.7.5.3.1.4.5, states that the AS-SIP – H.323 Gateway must support configuration of outbound voice call thresholds, inbound voice call thresholds, outbound video call thresholds, and inbound video call thresholds. The SUT does not support this conditional requirement.

(6) The UCR 2008 Change 3, section 5.3.2.7.5.3.1.4.6, states that the AS-SIP – H.323 Gateway must reject call requests received from the H.323 UC signaling platform that would exceed the appropriate [voice or video) call count threshold (or outbound call count threshold). The gateway must also reject initial routine INVITEs (i.e., not re-INVITEs) received from the WAN SS/MFSS that would exceed the appropriate (voice or video) call count threshold or inbound call count threshold. The AS-SIP – H.323 Gateway must either divert all precedence INVITEs to the attendant or the appropriate precedence INVITE to the attendant. The SUT met this requirement through testing.

(7) The UCR 2008 Change 3, section 5.3.2.7.5.3.1.4.7, states that each time the AS-SIP – H.323 Gateway receives a new voice call request, the AS-SIP – H.323 Gateway must conduct SAC as described in the subparagraphs below.

(a) If the initial INVITE received from the WAN SS/MFSS via the EBC is "routine" and the AS-SIP – H.323 Gateway is not enforcing directionalization, then one of the two following shall be used. The SUT met this requirement through testing.

1. If the voice call count is not at threshold, then increment the voice call count by one (the INVITE will be translated to H.323 signaling and sent to the H.323 UC signaling platform).

2. If the voice call count is at threshold then reject the INVITE.

(b) If the initial INVITE received from the WAN SS/MFSS is "routine" and the AS-SIP – H.323 Gateway is enforcing directionalization, then one of the two following shall be used. The SUT does not support this conditional requirement.

1. If the voice call count and inbound voice call count are not at threshold, then increment the voice call count by one and increment the inbound voice call count by one (the INVITE will be translated to H.323 signaling and sent to the H.323 UC signaling platform).

2. If either the voice call count or the inbound voice call count is at threshold, then reject the INVITE.

(c) If the initial INVITE received from the WAN SS/MFSS is a precedence INVITE and the AS-SIP – H.323 Gateway is not enforcing directionalization, then one of the two following shall be used.

1. If the AS-SIP – H.323 Gateway is configured to divert all precedence INVITEs to the attendant per requirement 5.3.2.7.5.1.4 a, then the INVITE is diverted to the attendant. The SUT met this requirement through testing.

2. If the AS-SIP – H.323 Gateway is configured to process the precedence INVITE per requirement 5.3.2.7.5.1.4b, then the requirements in the following subparagraphs apply. This conditional requirement does not apply to the SUT.

a. If the precedence INVITE would NOT cause the voice call count threshold to be exceeded, then increment the voice call count by one (the INVITE will be translated to H.323 signaling and sent to the H.323 UC signaling platform).

b. If the precedence INVITE would cause the voice call count threshold to be exceeded then divert the precedence INVITE to the attendant.

(d) If the initial INVITE received from the WAN SS/MFSS is a precedence INVITE and the AS-SIP – H.323 Gateway is enforcing directionalization, then one of the two requirements in the following subparagraphs apply. This conditional requirement does not apply to the SUT.

1. If the AS-SIP – H.323 Gateway is configured to divert all precedence INVITEs to the attendant per Requirement 5.3.2.7.5.1.4a, then the INVITE is diverted to the attendant.

2. If the AS-SIP – H.323 Gateway is configured to process the precedence INVITE per Requirement 5.3.2.7.5.1.4b, then:

a. If the precedence INVITE would NOT cause the voice call count threshold or the inbound voice call count threshold to be exceeded, then increment the voice call count by one and the inbound voice call count by one (the INVITE will be translated to H.323 signaling and sent to the H.323 UC signaling platform).

b. If the precedence INVITE would cause the voice call count threshold or inbound voice call count threshold to be exceeded, then divert the precedence INVITE to the attendant.

(e) If the call request is received from the H.323 UC signaling platform and the AS-SIP – H.323 Gateway is not enforcing directionalization, then one of the two following shall be used. The SUT met this requirement through testing.

1. If the voice call count is not at threshold, then increment the voice call count by one (the call request will be translated to an INVITE and sent to the WAN SS/MFSS).

2. If the voice call count is at threshold then reject the INVITE.

(f) If the call request is received from the H.323 UC signaling platform and the AS-SIP – H.323 Gateway is enforcing directionalization, then one of the two requirements in the following subparagraphs apply. This conditional requirement does not apply to the SUT.

1. If the voice call count and outbound voice call count are not at threshold, then increment the voice call count by one and the outbound voice call count by one (the call request will be translated to an INVITE and sent to the WAN SS/MFSS).

2. If either the voice call count or the outbound voice call count is at threshold, then reject the H.323 call request.

(8) The UCR 2008 Change 3, section 5.3.2.7.5.3.1.4.8, states that each time the AS-SIP – H.323 Gateway receives a new video session request, the AS-SIP – H.323 Gateway must conduct SAC as described in the subparagraphs below.

(a) If the initial INVITE received from the WAN SS/MFSS is “routine” and the AS-SIP – H.323 Gateway is not enforcing directionalization, then one of the two following shall be used. The SUT met this requirement through testing.

1. If the video call count is NOT at threshold and the video bandwidth in the INVITE request would not cause the video call count to exceed threshold, then increment the video call count by the appropriate number of VSUs (the INVITE will be translated to H.323 signaling and sent to the H.323 UC signaling platform).

2. If the video call count is at threshold or the video bandwidth in the INVITE request would cause the video call count to exceed threshold, then reject the INVITE.

(b) If the initial INVITE received from the WAN SS/MFSS is “routine” and the AS-SIP – H.323 Gateway is enforcing directionalization, then one of the two following shall be used. The SUT does not support this conditional requirement.

1. If the video call count and inbound video call count are NOT at threshold and the video bandwidth in the INVITE request would not cause the video call count or the inbound video call count to exceed threshold, then increment the video call

count and the inbound video call count by the appropriate number of VSUs (the INVITE will be translated to H.323 signaling and sent to the H.323 UC signaling platform).

2. If the video call count or inbound video call count is at threshold or the video bandwidth in the INVITE would cause the video call count or inbound video call count to exceed threshold, then reject the INVITE.

(c) If the initial INVITE received from the WAN SS/MFSS is a precedence INVITE and the AS-SIP – H.323 Gateway is not enforcing directionalization, then one of the two following shall be used.

1. If the AS-SIP – H.323 Gateway is configured to divert all precedence INVITEs to the attendant per Requirement 5.3.2.7.5.1.4a, then the INVITE is diverted to the attendant. The SUT met this requirement through testing.

2. If the AS-SIP – H.323 Gateway is configured to process the precedence INVITE per Requirement 5.3.2.7.5.1.4b, then the requirements in the following subparagraphs apply. This conditional requirement does not apply to the SUT.

a. If the video call count is NOT at threshold and the precedence INVITE would NOT cause the video call count threshold to be exceeded, then increment the video call count by the appropriate number of VSUs (the INVITE will be translated to H.323 signaling and sent to the H.323 UC signaling platform).

b. If the precedence INVITE would cause the video call count threshold to be exceeded, then divert the precedence INVITE to the attendant.

(d) If the initial INVITE received from the WAN SS/MFSS is a precedence INVITE and the AS-SIP – H.323 Gateway is enforcing directionalization, then one of the two requirements in the following subparagraphs apply. This conditional requirement does not apply to the SUT.

1. If the AS-SIP – H.323 Gateway is configured to divert all precedence INVITEs to the attendant per Requirement 5.3.2.7.5.1.4a, then the INVITE is diverted to the attendant.

2. If the AS-SIP – H.323 Gateway is configured to process the precedence INVITE per Requirement 5.3.2.7.5.1.4b, then:

a. If the video call count and inbound video call count are NOT at threshold and the precedence INVITE would NOT cause the video call count threshold or the inbound video call count threshold to be exceeded, then increment the video call count and the inbound video call count by the appropriate number of VSUs (the INVITE will be translated to H.323 signaling and sent to the H.323 UC signaling platform).

b. If the precedence INVITE would cause the video call count threshold or inbound video call count threshold to be exceeded, then divert the precedence INVITE to the attendant.

(e) If the call request is received from the H.323 UC signaling platform and the AS-SIP – H.323 Gateway is not enforcing directionalization, then one of the two following shall be used. The SUT met this requirement through testing.

1. If the video call count is not at threshold and the video bandwidth in the call request would not cause the video call count to exceed threshold then increment the video call count by the appropriate number of VSUs (the call request will be translated to an INVITE and sent to the WAN SS/MFSS).

2. If the video call count is at threshold or the video bandwidth in the call request would cause the video call count to exceed threshold then reject the call request.

(f) If the call request is received from the H.323 UC signaling platform and the AS-SIP – H.323 Gateway is enforcing directionalization, then one of the two requirements in the following subparagraphs apply. This conditional requirement does not apply to the SUT.

1. If the video call count and outbound video call count are not at threshold and the video bandwidth in the call request would not cause the video call count or the outbound video call count to exceed threshold, then increment the video call count and the outbound video call count by the appropriate number of VSUs (the call request will be translated to an INVITE and sent to the WAN SS/MFSS).

2. If either the video call count or the outbound video call count is at threshold or the video bandwidth in the call request would cause the video call count or outbound video call count to exceed threshold, then reject the INVITE.

c. AS-SIP – H.323 Gateway Management Function. The UCR 2008 Change 3, section 5.3.2.7.5.3.5, states that the AS-SIP – H.323 Gateway must meet the Generic Requirements (GRs) for NM function in the subparagraphs below.

(1) Section 5.3.2.17.1, Voice and Video Network Management Domain. This requirement was met with the vendor's LoC.

(2) Section 5.3.2.17.2, General Management Requirements. The AS-SIP – H.323 Gateway must support one pair of Ethernet management interfaces where one management interface is for communication with a local EMS and one management interface is for communication with a remote EMS. In addition, the AS-SIP – H.323 Gateway must support at least one additional Ethernet interface for carrying signaling and media streams for VVoIP traffic. This requirement was met with the vendor's LoC with the following minor exceptions. The SUT supports only one physical interface.

DISA adjudicated this as minor with the condition of fielding to be included in the vendor's deployment guide stipulating that local NM access and remote NM access will be provided via a single interface. The SUT meets the data reporting requirements; however, it does not use SNMP or XML to do so. DISA adjudicated this as minor with the condition of fielding to be included in the vendor's deployment guide stipulating an alternative procedure to manage the gateway.

(3) Section 5.3.2.17.3.1, Fault Management. This requirement was met with the vendor's LoC with the following minor exceptions. Fault reporting is not performed via SNMP. DISA adjudicated this as minor with the condition of fielding to be included in the vendor's deployment guide stipulating an alternative procedure to manage the gateway.

(4) Section 5.3.2.17.3.2.1, Read-Write Access to CM Data by the RTS EMS. This requirement was met with the vendor's LoC.

(5) Section 5.3.2.17.3.4.1, Near-Real-Time Network Performance Monitoring. This requirement was met with the vendor's LoC with the following minor exceptions. The vendor's LoC states Cisco is unable to verify the SUT capabilities for EMS interoperability; however, this capability is available using other existing UCM tools. DISA adjudicated this as minor with the condition of fielding to be included in the vendor's deployment guide stipulating an alternative procedure to manage the gateway.

(6) Section 5.3.2.17.3.4.2, Remote Network Management Commands (the LSC requirements apply to the AS-SIP – H.323 Gateway with the exception of Section 5.3.2.17.3.4.2.14, PEI/GEI Origination Capability Control). This requirement was met with the vendor's LoC with the following minor exceptions. The SUT supports only one physical interface. DISA adjudicated this as minor with the condition of fielding to be included in the vendor's deployment guide stipulating that local NM access and remote NM access will be provided via a single interface.

(7) Section 5.3.2.17.3.5, Security Management. This requirement was met with the vendor's LoC.

(8) Requirement 5.3.2.18.1, NM Requirements for CE Routers and EBCs. The vendor's LoC states the SUT meets the data reporting requirements referenced in this section; however the SUT does not use SNMP or XML to do so. DISA adjudicated this as minor with the condition of fielding to be included in the vendor's deployment guide stipulating an alternative procedure to manage the gateway.

(9) Section 5.3.2.18.2, Management Requirements for the ASAC (use these requirements for SAC only). This requirement was met with the vendor's LoC with the following minor exceptions. Call counts can be read from SUT management application; however, the vendor does not know if they can be read by VVoIP EMS. DISA adjudicated this as minor with the condition of fielding to be included in the vendor's deployment guide stipulating an alternative procedure to manage the gateway.

d. AS-SIP – H.323 Gateway-to-NMS Interface. The UCR 2008 Change 3, section 5.3.2.7.5.3.7, states that the AS-SIP – H.323 Gateway must provide an interface to the DISA NMS. The interface must consist of a 10/100-Mbps Ethernet connection as specified in Section 5.3.2.4.4. The SUT supports only one physical interface. DISA adjudicated this as minor with the DISA adjudicated this as minor with the condition of fielding to be included in the vendor's deployment guide stipulating that local NM access and remote NM access will be provided via a single interface.

e. IPv6 Requirements. The AS-SIP – H.323 Gateway must meet the IPv6 requirements for a Network Appliance/Simple Server (NA/SS) in Table 5.3.5-4. This requirement was met with the vendor's LoC with the following minor exceptions.

(1) The SUT meets the requirements for IPv4/IPv6 dual stack with the following minor exception. The SUT does not support dual stack for H.323. DISA adjudicated this as minor. There is no mature standard for dual stack with H.323 protocol. The SUT is certified for IPv4 only on the H.323 side of the gateway. The SUT will be fielded with IPv4 on the H.323 side of the gateway.

(2) The SUT VCS does not comply with MLD as described in RFC 2710. DISA adjudicated this as minor. The SUT will be fielded with IPv4 on the H.323 side of the gateway.

(3) The SUT partially complies with assigning the DSCP tagging requirements. The SUT VCS can be assigned any DSCP value 0-63, however this value is assigned to all types of packets to include OA&M, Signaling and Media. DISA accepted the vendor's Plan of Action and Milestones (PoA&M) and adjudicated this as minor.

(4) The SUT supports IPv6 Neighbor Discovery and SLAAC in accordance with RFCs 2461 and 2462 and not RFCs 4861 and 4862 as required by the UCR. During testing, there were no interoperability discrepancies related to IPv6 Neighbor Discovery or SLAAC. DISA accepted the vendor's PoA&M and adjudicated this as minor.

11.3 Information Assurance (IA). The UCR 2008 Change 3, section 5.3.2.7.5.3.3, states that the AS-SIP – H.323 Gateway must satisfy the IA requirements in Section 5.4 for a media gateway. Security testing is accomplished via DISA-led IA test teams and published in a separate report, Reference (e).

11.4 Other. None.

12. TEST AND ANALYSIS REPORT. No detailed test report was developed in accordance with the Program Manager's request. JITC distributes interoperability information via the JITC Electronic Report Distribution (ERD) system, which uses Unclassified-But-Sensitive Internet Protocol Router Network (NIPRNet) e-mail. More comprehensive interoperability status information is available via the JITC System Tracking Program (STP). STP is accessible by .mil/gov users on the NIPRNet at https://stp.fhu.disa.mil. Test reports, lessons learned, and related testing documents and references are on the JITC Joint Interoperability Tool (JIT) at http://jit.fhu.disa.mil (NIPRNet). Due to the sensitivity of the information, the Information Assurance Accreditation Package (IAAP) that contains the approved configuration and deployment guide must be requested directly through government civilian or uniformed military personnel from the Unified Capabilities Certification Office (UCCO), e-mail: disa.meade.ns.list.unified-capabilities-certification-office@mail.mil. All associated data is available on the DISA UCCO website located at website located at http://www.disa.mil/Services/Network-Services/UCCO.

SYSTEM FUNCTIONAL AND CAPABILITY REQUIREMENTS

The Assured Services Session Initiation Protocol (AS-SIP) to H.323 Gateway components have required and conditional features and capabilities that are established by the Unified Capabilities Requirements (UCR) 2008, Change 3, Section 5.3.2.7.5. The system under test does not need to provide conditional requirements. However, if a capability is provided, it must function according to the specified requirements. The detailed Functional requirements (FR) and Capability Requirements (CR) for AS-SIP to H.323 Gateways are listed in Table 3-1. Detailed Information Assurance (IA) requirements are included in Reference (e).

The Government has adopted RFC 4123 – Session Initiation Protocol (SIP) – H.323 Interworking Requirements as the document which describes the requirements for the AS-SIP – H.323 Gateway. Internet draft-agrawal-sip-h323-interworking-01.txt is cited as guidance to be used in implementing the AS-SIP – H.323 Gateway. The requirements in Table 3-1 are additional Government requirements. The AS-SIP – H.323 Gateway is not an assured services appliance because H.323 is not an assured services protocol, and its placement in this section is for requirements grouping purposes and should not be interpreted as implying that the AS-SIP – H.323 Gateway is an Assured Services appliance. The AS-SIP – H.323 Gateway is a standalone system for testing purposes.

Table 3-1. AS-SIP to H.323 Gateway CRs and FRs

ID	Requirement	R/C	UCR Reference
1	The AS-SIP – H.323 Gateway must implement call count thresholds for voice sessions and for video sessions in order to perform Session Admission Control (SAC). See Section 5.3.2.7.5.3.1.4, Session Admission Control, for more details.	R	5.3.2.7.5.1.1
2	When the AS-SIP – H.323 Gateway receives a call request from the H.323 UC signaling platform then the AS-SIP – H.323 Gateway must: a. Check the appropriate (voice or video) call count (and outbound call count in the case of directionalization) to determine whether there are available bandwidth resources to support the call request. b. If the new call request would not exceed the appropriate (voice or video) call count threshold (and outbound call count threshold) then the AS-SIP – H.323 Gateway interworks the call request by: (1) Incrementing the call count (and outbound call count in the case of directionalization). (2) Generating a "routine" level AS-SIP INVITE that advertises equivalent capabilities to those specified in the received call request. (3) Adding a CCA-ID parameter to the Contact header. (4) Adding a route set comprising two Route headers where the first Route header is the SIP URI for the EBC at the enclave, and the second Route header is the SIP URI for the EBC serving the WAN SS/MFSS. (5) Forwarding the INVITE message to the EBC at the enclave. c. If the appropriate (voice or video) call count (or outbound call count) is at threshold or the call request would cause the AS-SIP – H.323 Gateway to exceed the call count threshold (or outbound call count threshold) then the AS-SIP – H.323 Gateway must reject the call.	R	5.3.2.7.5.1.2

Table 3-1. AS-SIP to H.323 Gateway CRs and FRs (continued)

ID	Requirement (See note.)	R/C	UCR Reference
3	When an AS-SIP – H.323 Gateway receives an initial routine AS-SIP INVITE (i.e., not a re-INVITE) from the WAN SS/MFSS (via the EBC), then the AS-SIP – H.323 Gateway must: a. Check the appropriate (voice or video) call count (and inbound call count in the case of directionalization) to determine whether there are available bandwidth resources to support the call request. b. If the new call request would not exceed the appropriate (voice or video) call count threshold (and inbound call count threshold) then the AS-SIP – H.323 Gateway increments the appropriate (voice or video) call count (and inbound call count) and interworks the INVITE to H.323. c. If the appropriate (voice or video) call count (or inbound call count) is at threshold or the call request would cause the AS-SIP – H.323 Gateway to exceed the appropriate (voice or video) call count threshold (or inbound call count threshold) then the AS-SIP – H.323 Gateway must reject the call.	R	5.3.2.7.5.1.3
4	The AS-SIP – H.323 Gateway must support the following two methods for processing initial precedence AS-SIP INVITEs received from the WAN SS/MFSS via the EBC and the choice of method must be software configurable: a. Upon receipt of the initial precedence AS-SIP INVITE request the AS-SIP – H.323 Gateway diverts the precedence INVITE to the attendant, or b. Upon receipt of the initial precedence AS-SIP INVITE request, the AS-SIP – H.323 Gateway determines whether the appropriate (voice or video) call count (or inbound call count in the case of directionalization) is at threshold or whether the call request would cause the AS-SIP – H.323 Gateway to exceed the appropriate (voice or video) call count threshold or inbound call count threshold: (1) If the precedence AS-SIP INVITE would cause the appropriate (voice or video) call count threshold (or inbound call count threshold) to be exceeded, then the precedence AS-SIP INVITE is forwarded to the attendant. NOTE: The AS-SIP – H.323 Gateway must NOT conduct preemption on behalf of an inbound precedence AS-SIP INVITE. (2) If the precedence AS-SIP INVITE would NOT cause the appropriate call count threshold (or inbound call count threshold) to be exceeded, then the AS-SIP – H.323 Gateway treats the inbound precedence AS-SIP INVITE request as if it were a routine inbound call request and increments the appropriate (voice or video) call count (and inbound call count) and interworks the INVITE to platform.	R	5.3.2.7.5.1.4
5	Table 5.3.2.7-7, AS-SIP – H.323 Gateway support for VoIP and Video Signaling Interfaces, provides a complete list of the AS-SIP – H.323 Gateway signaling requirements.	R	5.3.2.7.5.3.1
6	The CCA is part of the SCS functions and includes the IWF (signaling) function. The scope of these CCA requirements covers the following areas: 1. AS-SIP signaling protocol implementation for voice and video calls 2. H.323 signaling protocol implementation for voice and video calls (where signaling protocol implementation refers to the signaling being used by the H.323 UC signaling platform) 3. Control of sessions within the AS-SIP – H.323 Gateway including: a. H.323 sessions between the AS-SIP – H.323 Gateway and the H.323 UC signaling platform b. AS-SIP sessions between the AS-SIP – H.323 Gateway and the serving WAN SS/MFSS 4. Support for interactions with other network appliance functions including: a. Admission control b. Information Assurance c. Media interworking d. Appliance Management functions.	R	5.3.2.7.5.3.1.1
7	The role of the IWF within the CCA is to 1. Interwork the messages of the H.323 VoIP signaling protocol into AS-SIP signaling messages. 2. Interwork AS-SIP signaling messages into messages of the H.323 VoIP signaling protocol. The CCA IWF must support the AS-SIP consistent with the detailed AS-SIP requirements in Section 5.3.4, AS-SIP Requirements. The CCA IWF must secure the AS-SIP protocol using TLS, as described in Section 5.4, Information Assurance Requirements. The CCA IWF component of the AS-SIP – H.323 Gateway must ensure that when the supplementary services enumerated in the UCR (i.e., Call Hold, Call Waiting, Precedence Call Waiting, Call Forwarding, Call Transfer) are performed by a served H.323 UC signaling platform that the AS-SIP – H.323 Gateway presents UCR-compliant call flows to the signaling appliances in the UC network per UCR Section 5.3.4.13.	R	5.3.2.7.5.3.1.1.1

Table 3-1. AS-SIP to H.323 Gateway CRs and FRs (continued)

ID	Requirement (See note.)	R/C	UCR Reference
8	The AS-SIP – H.323 Gateway does NOT conduct preemption. Whenever the AS-SIP – H.323 Gateway receives a H.323 signaling message from the H.323 UC signaling platform that translates it into an INVITE, UPDATE, or REFER request, then the AS-SIP – H.323 Gateway must generate a Resource-Priority header having a ROUTINE priority level IAW Section 5.3.4.10.2 Precedence Level Communicated over SIP Signaling. Whenever the AS-SIP – H.323 Gateway receives an INVITE, UPDATE, or REFER request from the WAN SS/MFSS via the EBC, then the AS-SIP – H.323 Gateway must process the Resource-Priority header to distinguish a ROUTINE call from a precedence call. In the case of a ROUTINE call the AS-SIP – H.323 Gateway follows the procedure in UCR Requirement 5.3.2.7.5.1.3. In the case of a precedence call, the AS-SIP – H.323 Gateway follows the procedure in UCR Requirement 5.3.2.7.5.1.4.	R	5.3.2.7.5.3.1.2
9	When the AS-SIP – H.323 Gateway receives a call request from the H.323 UC signaling platform, then the AS-SIP – H.323 Gateway must map the telephony numbers received from the initial H.323 signaling message to SIP URIs IAW Section 5.3.4.14.3, SIP URI and Mapping of Telephony Number into SIP URI, and UCR Section 5.3.4.7.6, SIP URI and Mapping of Telephone Number into SIP URI.	R	5.3.2.7.5.3.1.3
10	The AS-SIP – H.323 Gateway must conduct SAC as detailed in this section in place of the ASAC required of LSCs.		5.3.2.7.5.3.1.4.1
11	The AS-SIP – H.323 Gateway must support directionalization. NOTE: Whenever the H.323 UC signaling platform supports Directionalization, then directionalization will be performed in the H.323 UC signaling platform and not in the AS-SIP – H.323 Gateway. (C)	R	5.3.2.7.5.3.1.4.2
12	The AS-SIP – H.323 Gateway must support code blocking. NOTE: Whenever the H.323 UC signaling platform supports code blocking then code blocking will be performed in the H.323 UC signaling platform and not in the AS-SIP – H.323 Gateway.	R	5.3.2.7.5.3.1.4.3
13	The AS-SIP – H.323 Gateway must support configuration of total voice call thresholds and total video call thresholds.	R	5.3.2.7.5.3.1.4.4
14	The AS-SIP – H.323 Gateway must support configuration of outbound voice call thresholds, inbound voice call thresholds, outbound video call thresholds, and inbound video call thresholds.	R	5.3.2.7.5.3.1.4.5
15	Session Admission Control refers to the enforcement of voice and video session thresholds whereby the AS-SIP – H.323 Gateway must: a. Reject call requests received from the H.323 UC signaling platform that would exceed the appropriate [voice or video) call count threshold (or outbound call count threshold). b. Reject initial routine INVITEs (i.e., not re-INVITEs) received from the WAN SS/MFSS that would exceed the appropriate (voice or video) call count threshold or inbound call count threshold. c. Per Requirement 5.3.2.7.5.1.4, depending on the desired software configuration of the given AS-SIP – H.323 Gateway either implement Requirement 5.3.2.7.5.1.4 a to divert all precedence INVITEs to the attendant or implement Requirement 5.3.2.7.5.1.4 b(1) if the INVITE would cause the appropriate (voice or video) call count threshold (or inbound call count threshold) to be exceeded and divert the precedence INVITE to the attendant.	R	5.3.2.7.5.3.1.4.6

Table 3-1. AS-SIP to H.323 Gateway CRs and FRs (continued)

ID	Requirement (See note.)	R/C	UCR Reference
16	Each time the AS-SIP – H.323 Gateway receives a new voice call request the AS-SIP – H.323 Gateway must conduct SAC as follows: a. If the initial INVITE received from the WAN SS/MFSS via the EBC is “routine” and the AS-SIP – H.323 Gateway is not enforcing directionalization, (1) If the voice call count is not at threshold, then increment the voice call count by one (the INVITE will be translated to H.323 signaling and sent to the H.323 UC signaling platform). (2) If the voice call count is at threshold then reject the INVITE. b. If the initial INVITE received from the WAN SS/MFSS is “routine” and the AS-SIP – H.323 Gateway is enforcing directionalization, then: (1) If the voice call count and inbound voice call count are not at threshold, then increment the voice call count by one and increment the inbound voice call count by one (the INVITE will be translated to H.323 signaling and sent to the H.323 UC signaling platform). (2) If either the voice call count or the inbound voice call count is at threshold, then reject the INVITE. c. If the initial INVITE received from the WAN SS/MFSS is a precedence INVITE and the AS-SIP – H.323 Gateway is not enforcing directionalization, then: (1) If the AS-SIP – H.323 Gateway is configured to divert all precedence INVITEs to the attendant per Requirement 5.3.2.7.5.1.4 a, then the INVITE is diverted to the attendant. (2) If the AS-SIP – H.323 Gateway is configured to process the precedence INVITE per Requirement 5.3.2.7.5.1.4b, then: (a) If the precedence INVITE would NOT cause the voice call count threshold to be exceeded, then increment the voice call count by one (the INVITE will be translated to H.323 signaling and sent to the H.323 UC signaling platform). (b) If the precedence INVITE would cause the voice call count threshold to be exceeded then divert the precedence INVITE to the attendant. d. If the initial INVITE received from the WAN SS/MFSS is a precedence INVITE and the AS-SIP – H.323 Gateway is enforcing directionalization, then: (1) If the AS-SIP – H.323 Gateway is configured to divert all precedence INVITEs to the attendant per Requirement 5.3.2.7.5.1.4a, then the INVITE is diverted to the attendant (2) If the AS-SIP – H.323 Gateway is configured to process the precedence INVITE per Requirement 5.3.2.7.5.1.4b, then (a) If the precedence INVITE would NOT cause the voice call count threshold or the inbound voice call count threshold to be exceeded, then increment the voice call count by one and the inbound voice call count by one (the INVITE will be translated to H.323 signaling and sent to the H.323 UC signaling platform). (b) If the precedence INVITE would cause the voice call count threshold or inbound voice call count threshold to be exceeded, then divert the precedence INVITE to the attendant. e. If the call request is received from the H.323 UC signaling platform and the AS-SIP – H.323 Gateway is not enforcing directionalization, then: (1) If the voice call count is not at threshold, then increment the voice call count by one (the call request will be translated to an INVITE and sent to the WAN SS/MFSS). (2) If the voice call count is at threshold then reject the INVITE. f. If the call request is received from the H.323 UC signaling platform and the AS-SIP – H.323 Gateway is enforcing directionalization, then: (1) If the voice call count and outbound voice call count are not at threshold, then increment the voice call count by one and the outbound voice call count by one (the call request will be translated to an INVITE and sent to the WAN SS/MFSS). (2) If either the voice call count or the outbound voice call count is at threshold, then reject the H.323 call request.		5.3.2.7.5.3.1.4.7

Table 3-1. AS-SIP to H.323 Gateway CRs and FRs (continued)

ID	Requirement (See note.)	R/C	UCR Reference
17	Each time the AS-SIP – H.323 Gateway receives a new video session request, then the AS-SIP – H.323 Gateway must conduct SAC as follows: a. If the initial INVITE received from the WAN SS/MFSS is "routine" and the AS-SIP – H.323 Gateway is not enforcing directionalization, then: (1) If the video call count is NOT at threshold and the video bandwidth in the INVITE request would not cause the video call count to exceed threshold, then increment the video call count by the appropriate number of VSUs (the INVITE will be translated to H.323 signaling and sent to the H.323 UC signaling platform). (2) If the video call count is at threshold or the video bandwidth in the INVITE request would cause the video call count to exceed threshold, then reject the INVITE. b. If the initial INVITE received from the WAN SS/MFSS is "routine" and the AS-SIP – H.323 Gateway is enforcing directionalization, then: (1) If the video call count and inbound video call count are NOT at threshold and the video bandwidth in the INVITE request would not cause the video call count or the inbound video call count to exceed threshold, then increment the video call count and the inbound video call count by the appropriate number of VSUs (the INVITE will be translated to H.323 signaling and sent to the H.323 UC signaling platform). (2) If the video call count or inbound video call count is at threshold or the video bandwidth in the INVITE would cause the video call count or inbound video call count to exceed threshold, then reject the INVITE. c. If the initial INVITE received from the WAN SS/MFSS is a precedence INVITE and the AS-SIP – H.323 Gateway is not enforcing directionalization, then: (1) If the AS-SIP – H.323 Gateway is configured to divert all precedence INVITEs to the attendant per Requirement 5.3.2.7.5.1.4a, then the INVITE is diverted to the attendant. (2) If the AS-SIP – H.323 Gateway is configured to process the precedence INVITE per Requirement 5.3.2.7.5.1.4b, then: (a) If the video call count is NOT at threshold and the precedence INVITE would NOT cause the video call count threshold to be exceeded, then increment the video call count by the appropriate number of VSUs (the INVITE will be translated to H.323 signaling and sent to the H.323 UC signaling platform). (b) If the precedence INVITE would cause the video call count threshold to be exceeded, then divert the precedence INVITE to the attendant. d. If the initial INVITE received from the WAN SS/MFSS is a precedence INVITE and the AS-SIP – H.323 Gateway is enforcing directionalization, then: (1) If the AS-SIP – H.323 Gateway is configured to divert all precedence INVITEs to the attendant per Requirement 5.3.2.7.5.1.4a, then the INVITE is diverted to the attendant. (2) If the AS-SIP – H.323 Gateway is configured to process the precedence INVITE per Requirement 5.3.2.7.5.1.4b, then: (a) If the video call count and inbound video call count are NOT at threshold and the precedence INVITE would NOT cause the video call count threshold or the inbound video call count threshold to be exceeded, then increment the video call count and the inbound video call count by the appropriate number of VSUs (the INVITE will be translated to H.323 signaling and sent to the H.323 UC signaling platform). (b) If the precedence INVITE would cause the video call count threshold or inbound video call count threshold to be exceeded, then divert the precedence INVITE to the attendant. e. If the call request is received from the H.323 UC signaling platform and the AS-SIP – H.323 Gateway is not enforcing directionalization, then: (1) If the video call count is not at threshold and the video bandwidth in the call request would not cause the video call count to exceed threshold then increment the video call count by the appropriate number of VSUs (the call request will be translated to an INVITE and sent to the WAN SS/MFSS). (2) If the video call count is at threshold or the video bandwidth in the call request would cause the video call count to exceed threshold then reject the call request. f. If the call request is received from the H.323 UC signaling platform and the AS-SIP – H.323 Gateway is enforcing directionalization, then: (1) If the video call count and outbound video call count are not at threshold and the video bandwidth in the call request would not cause the video call count or the outbound video call count to exceed threshold, then increment the video call count and the outbound video call count by the appropriate number of VSUs (the call request will be translated to an INVITE and sent to the WAN SS/MFSS). (2) If either the video call count or the outbound video call count is at threshold or the video bandwidth in the call request would cause the video call count or outbound video call count to exceed threshold, then reject the INVITE.		5.3.2.7.5.3.1.4.8

Table 3-1. AS-SIP to H.323 Gateway CRs and FRs (continued)

<table>
<tr><th>ID</th><th colspan="2">Requirement (See note.)</th><th>R/C</th><th>UCR Reference</th></tr>
<tr><td>18</td><td colspan="2">Summary of Relevant Media Packet Requirements from other UCR Sections:
The AS-SIP – H.323 Gateway must support the audio Codecs in Section 5.3.2.6.1.2, Video Audio Codecs.
The AS-SIP – H.323 Gateway must comply with Section 5.3.2.6.1.4, Voice over IP Sampling Standard, for the sampling rates.
The AS-SIP – H.323 Gateway must support the audio and video Codecs as specified in Section 5.3.2.6.2.2, Video Codecs (Including Associated Audio Codecs).</td><td>R</td><td>5.3.2.7.5.3.2</td></tr>
<tr><td>19</td><td colspan="2">The AS-SIP – H.323 Gateway must meet IA requirements IAW UCR 2008, Change 3, Section 5.4 for a media gateway.</td><td>R</td><td>5.3.2.7.5.3.3</td></tr>
<tr><td>20</td><td colspan="2">Upon total loss of WAN transport the AS-SIP – H.323 Gateway becomes incapable of exchanging signaling messages between the connected H.323 UC signaling platform and the UC WAN and incapable of exchanging interworked media packets between the EIs served by the H.323 UC signaling platform and the UC WAN. The immediate consequence is that the users on the existing voice and video sessions can no longer successfully send or receive media, and will go on-hook. The signaling termination messages (triggered by going on-hook) will fail to transit the WAN due to the loss of WAN transport. In addition, since the AS-SIP – H.323 Gateway provides the only connectivity to the UC WAN for the H.323 UC signaling platform, the H.323 UC signaling platform loses the ability to establish new calls over the UC WAN until WAN connectivity is restored.</td><td></td><td>5.3.2.7.5.3.4</td></tr>
<tr><td rowspan="17">21</td><td rowspan="17">AS-SIP – H.323 Gateway Manage-ment Function</td><td>Section 5.3.2.17.1, Voice and Video Network Management Domain</td><td>R</td><td rowspan="17">5.3.2.7.5.3.5</td></tr>
<tr><td>Section 5.3.2.17.2, General Management Requirements</td><td></td></tr>
<tr><td>The AS-SIP – H.323 Gateway must support one pair of Ethernet management interfaces where one management interface is for communication with a local EMS and one management interface is for communication with a remote EMS. In addition, the AS-SIP – H.323 Gateway must support at least one additional Ethernet interface for carrying signaling and media streams for VVoIP traffic.</td><td></td></tr>
<tr><td>Section 5.3.2.17.3.1, Fault Management</td><td>R</td></tr>
<tr><td>Section 5.3.2.17.3.2.1, Read-Write Access to CM Data by the RTS EMS</td><td></td></tr>
<tr><td>Section 5.3.2.17.3.4.1, Near-Real-Time Network Performance Monitoring</td><td>R</td></tr>
<tr><td>Section 5.3.2.17.3.4.2, Remote Network Management Commands (the LSC requirements apply to the AS-SIP – H.323 Gateway with the exception of Section 5.3.2.17.3.4.2.14, PEI/GEI Origination Capability Control)</td><td></td></tr>
<tr><td>Section 5.3.2.17.3.5, Security Management</td><td></td></tr>
<tr><td>Section 5.3.2.17.4, Data Classification</td><td></td></tr>
<tr><td>Section 5.3.2.17.5, Management of Appliance Software</td><td>R</td></tr>
<tr><td>Requirement 5.3.2.18.1, NM Requirements for CE Routers and EBCs</td><td></td></tr>
<tr><td>Section 5.3.2.18.2, Management Requirements for the ASAC (use these requirements for SAC only)</td><td>R</td></tr>
<tr><td>Section 5.3.2.18.3.1.1, CCA Support for Capacity Installation, but not including Section 5.3.2.18.3.1.1.1, MG-Related Configuration, and Section 5.3.2.18.3.1.1.2, SG-Related Data)</td><td>R</td></tr>
<tr><td>Section 5.3.2.18.3.3, CCA Support for Fault Localization</td><td>R</td></tr>
<tr><td>Section 5.3.2.18.3.4, CCA Support for Testing</td><td>R</td></tr>
<tr><td>25</td><td colspan="2">The AS-SIP – H.323 Gateway must provide an interface to the DISA NMS. The interface must consist of a 10/100-Mbps Ethernet connection as specified in Section 5.3.2.4.4, VVoIP NMS Interface Requirements.</td><td>R</td><td>5.3.2.7.5.3.7</td></tr>
<tr><td>26</td><td colspan="2">The AS-SIP – H.323 Gateway shall meet the product quality factors specified in Section 5.3.2.5.2, Product Quality Factors.</td><td>R</td><td>5.3.2.7.5.3.8</td></tr>
<tr><td>27</td><td colspan="2">IPv6 Product Requirements for NA/SS</td><td>R</td><td>5.3.5.4</td></tr>
</table>

Table 3-1. AS-SIP to H.323 Gateway CRs and FRs (continued)

LEGEND:

ASAC	Assured Services Admission Control
AS-SIP	Assured Services Session Initiation Protocol
C	Conditional
CCA	Call Connection Agent
CE	Customer Edge
CM	Configuration Management
CR	Capability Requirement
DISA	Defense Information Systems Agency
EBC	Edge Boundary Controller
EI	End Instrument
EMS	Element Management System
FR	Functional Requirement
GEI	Generic End Instrument
IA	Information Assurance
IAW	in accordance with
ID	Identification
IPv6	Internet Protocol version 6
IWF	Inter-working Function
H.323	Standard for multi-media communications on packet-based networks
LSC	Local Session Controller
Mbps	Megabits per second
MG	Media Gateway
MFSS	Multi-function Softswitch
NA/SS	Network Appliance/Simple Server
NM	Network Management
NMS	Network Management System
PEI	Proprietary End Instrument
R	Required
RTS	Real Time Services
SAC	Session Admission Control
SCS	Session Control and Signaling
SG	Signaling Gateway
SIP	Session Initiation Protocol
SS	Softswitch
TLS	Transport Layer Security
UC	Unified Capabilities
UCR	Unified Capabilities Requirements
URI	Uniform Resource Indicator
VoIP	Voice over Internet Protocol
VVoIP	Voice and Video over Internet Protocol
WAN	Wide Area Network